3 Network Dialing Schemes

Introduction

This chapter contains information on selecting and configuring Alcatel VoIP Network Dialing Schemes (AVNDS) which are used to translate dialed digits into IP addresses on the switch. At least one dialing scheme must be configured to support a Voice over IP network.

The dialing scheme examples discussed in this chapter are daughtercard centric, and typically consist of two PBXs with corresponding voice switching daughtercards each connected by one incoming only and one outgoing only trunk. In most cases the PBX is assumed to be trunked to the North American PSTN (Public Switched Telephone Network); however, some examples have voice daughtercards connected to the PSTN. It should be presumed also that all calls going to the PSTN are directed by Telco Central Offices. The WAN links between the switches (or some other device) are via the WSM or WSX modules which provide the ports, e.g., T1, E1, for data communications. The voice daughtercards (VSD, VSB and VSA) provide the telephony ports, e.g., T1, E1, Euro ISDN BRI, FXO and FXS for voice communications.

Variations to the dialing scheme configurations entail other likely scenarios in a VoIP network, including the use of hunt groups, site prefixes, strip digits, fax over IP and caller ID. Dialing schemes for special configurations, such as using VoIP in the switch with the OmniPCX 4400, are provided as well. All dialing schemes can be used in OmniAccess 512 and Omni Switch/Router configurations.

To simplify the configuration process, a VSM (Voice Switching Module) partial text-based ASCII configuration boot file (**vsmboot.asc**) has been created for each dialing scheme. Each partial boot file contains the specific CLI commands needed to implement a selected dialing scheme, and should be merged with the complete master boot file (**vsmboot_master.asc**), modified accordingly and then installed on the switch. Refer to Chapter 4, “Setup and Installation,” for further details and an example boot file configuration. For specific details on the VoIP text-based command line interface (CLI) commands relative to the boot files and dialing schemes, see Chapter 5, “VoIP Commands”.

Tbd - The table below lists dialing plans that use particular VoIP features. The table on the following page lists each of the dialing scheme examples, and contains decision criteria for determining the most suitable dialing scheme to use. Examples and descriptions of the dialing schemes in this chapter are intended to serve as guidelines in the development of enterprise-specific network VoIP dialing schemes.

VoIP Feature	Dialing Scheme	CLI Reference
H.323 gateway to voice daughtercard **(A)**	1, 11, 9, tbd	page 222
H.323 gateway to H.323 gatekeeper (RADVision) **(B)**	tbd	page 217
H.323 gateway to Microsoft NetMeeting (without FastStart) **(C)**	tbd	pages 20, 21
Local channel — 48 individual hunt groups (One channel per group) **(D)**	tbd	pages 222, 225
Local channel — four hunt groups (12 T1 channels per group) **(E)**	tbd	pages 222, 225
Local channel — two hunt groups (24 T1 channels per group) **(F)**	tbd	pages 222, 225
Local channel — one hunt group (48 channels across two T1s) **(G)**	tbd	pages 222, 225
Local channel — one hunt group (60 channels across two E1s) **(H)**	tbd	pages 222, 225
Site prefix — no site prefix **(I)**	1, tbd	page 231, 232
Site prefix — single digit **(J)**	2, 3, tbd	page 231, 232
Site prefix — multiple digits **(J1)**	tbd	page 231, 232
Voice phone group type — three digit local extensions **(K)**	1, 9, tbd	page 233, 236, 239
Voice phone group type — four digit local extensions **(L)**	1, 11, tbd	page 233, 236, 239
Voice phone group type — eleven digit local extensions **(M)**	1, 9, tbd	pages 233, 239
Voice phone group type — NANP extensions **(N)**	tbd	pages 233, 239
Voice phone group type — INTL extension **(O)**	tbd	page 233
Voice phone group type — PSTN NANP **(P)**	tbd	page 233
Voice phone group type — PSTN International (INTL) **(Q)**	1, 9, tbd	page 233
Strip digit length — no strip digits **(R)**	11, tbd	page 240
Strip digit length — 1 **(S)**	tbd	page 240
Strip digit length — 2 **(T)**	tbd	page 240
Strip digit length — 4 **(U)**	tbd	page 240
Strip digit length — 7 **(V)**	tbd	page 240

How to Select a Dialing Scheme

Use the decision criteria in the far right column of the table below to determine the most appropriate dialing scheme to follow when configuring the network for VoIP in the switch. The dialing schemes are discussed in this chapter in numerical order, but are categorized into three distinct types:

- **VoIP Networks without PSTN** (Dialing Schemes 1-12)

  Dialing scheme examples in this group do not connect to the PSTN. It is assumed that the PBX handles the routing of the call to the VoIP network. The first two examples are considered basic dialing schemes, while the remaining examples in this group demonstrate more complex VoIP dialing scheme concepts, such as how to use hunt groups (to multiply and split T1 lines), strip digits, or an H.323 gatekeeper.

- **VoIP Networks with PSTN** (Dialing Schemes 13-18)

  Dialing scheme examples in this group connect the voice daughtercards to the North American PSTN, and cover the use of strip digits, fax over IP, and caller ID (forwarding and static). *International (ISDN) PSTN and Caller ID Forwarding not available this release.*

- **VoIP Networks with Interoperability** (Dialing Schemes 19-24)

  Dialing scheme examples in this group allow VoIP networks to work with other functionally related equipment inlcuding H.323 Gateways, the OmniPCX 4400 and assorted PBXs.

All dialing schemes in this chapter can be modified to be used with the VSD, VSB and VSA voice switching daughtercards with the following exceptions:

- Dialing schemes No. 7 and 8 (Fractional T1 Hunt Groups) apply only to VSD and VSB daughtercards. Fractional type hunt groups do not apply to VSAs because analog channels can only be combined, not multiplied or split.
- Dialing schemes No. 17 and 18 (Caller ID, Forwarding and Static) apply only to VSD E1 or VSA daughtercards, or VSD T1 when configured for FXS Loop Start signaling.
- Except when local channels are used, all AVNDS commands function with the H.323 endpoints, e.g., OmniPCX, Cisco Routers, Microsoft NetMeeting.

No.	Dialing Scheme Examples / VoIP Networks without PSTN	Decision Criteria
1	Four Digit Extensions and Two Voice Daughtercards	Basic VoIP Network
2	Four Digit Extensions and Three Voice Daughtercards	Expanded VoIP Network
3	Hunt Groups — One Hunt Group (48 channels across two T1s)	One Hunt Group Per T1 Voice Daughtercard
4	Hunt Groups — One Hunt Group (60 channels across two E1s)	One Hunt Group Per E1 Voice Daughtercard
5	Hunt Groups — One Hunt Group (96 channels across four T1s)	One Hunt Group Across Two Voice Daughtercards
6	Hunt Groups — One Hunt Group (144 channels across six T1s)	One Hunt Group Across Three Voice Daughtercards
7	Hunt Groups — Four Hunt Groups (12 channels per group)	Fractional T1 Hunt Groups
8	Hunt Groups — 48 Invidual Hunt Groups (One channel per group)	Fractional T1 Hunt Groups
9	Strip Digits — Trunk Groups and Mixed Length Extensions	Unique mixed length extensions
10	Strip Digits — Trunk Groups and Three Digit Extensions	Common extensions, Unique, two digit site prefix
11	Strip Digits — Trunk Groups and Four Digit Extensions	Common extensions Unique, one digit site prefix
tbd	Strip Digits — One Trunk Group and Eleven Digit Extensions	Unique NANP extensions
12	H.323 Gatekeeper	Unique extensions Complex VoIP Network with H.323 gatekeeper

No.	Dialing Scheme Examples / VoIP Networks with PSTN	Decision Criteria
13	No. American PSTN — Four Digit Extensions and Direct Inward Dial (DID)	Unique NANP extensions Unique Site Prefix
14	Intl. (ISDN) PSTN — Four Digit Extensions and Direct Inward Dial (DID).	No Amer and Intl. Sites
15	No. American PSTN — Strip Digit Length (4) (tbd-to be replaced)	to be replaced
16	No. American PSTN — Fax over IP Network	Toll-Saving Fax Calls
17	No. American PSTN — Caller ID (Forwarding). *Not available this release.*	tbd
18	No American PSTN — Caller ID (Static).	Analog Voice Daughtercard

No.	Dialing Scheme Examples / VoIP Networks with Interoperability	Decision Criteria
19	H.323 Gateway — Microsoft NetMeeting (w/o FastStart)	VoIP to 3rd Party H.323 Software
20	H.323 Gateway — Cisco Routers	VoIP to 3rd Party H.323 Hardware
21	H.323 Gateway — OmniPCX 4400	VoIP to OmniPCX LIOE card
22	Omni PCX 4400 — E1 QSIG	Interoperating via E1 QSIG
23	Omni PCX 4400 — Euro PRI	Interoperating via Euro PRI
24	Other PBXs — T1	Interoperating with 3rd Party PBX

Tbd — dialing scheme examples not yet updated to match trunk list below.

The trunk list below is provided as a general reference guideline to each of the diagrams used in the dialing scheme examples. Note that most examples use only one or two PBXs, and one or two voice daughtercards.

Lines **(A)**: Telephone/Fax lines off of PBX #1.

Lines **(B)**: Telephone/Fax lines off of PBX #2.

Lines **(C)**: Telephone/Fax lines off of PBX #3.

Trunk **(D)**: Inbound trunk to PBX #1.

Trunk **(E)**: Inbound trunk to PBX #2.

Trunk **(F)**: Inbound trunk to PBX #3.

Trunk **(G)**: Inbound trunk to Voice Daughtercard #1.

Trunk **(H)**: Inbound trunk to Voice Daughtercard #2.

Trunk **(I)**: Inbound trunk to Voice Daughtercard #3.

Trunks **(J)**, **(K)**: Inbound/Outbound trunk (to Voice Daughtercard #1).

Trunks **(L)**, **(M)**: Inbound/Outbound trunk (to Voice Daughtercard #2).

Trunks **(N)**, **(O)**: Inbound/Outbound trunk (to Voice Daughtercard #3).

Trunks **(P)**, **(Q)**: Inbound/Outbound trunk (to Voice Daughtercard #4).

Trunk **(R)**: Inbound/Outbound trunk (between Voice Daughtercard #1 and PSTN)

Trunk **(S)**: Inbound/Outbound trunk (between Voice Daughtercard #2 and PSTN)

Trunk **(T)**: Inbound/Outbound trunk (between Voice Daughtercard #3 and PSTN)

Trunk **(U)**: Inbound/Outbound trunk (between Voice Daughtercard #4 and PSTN)

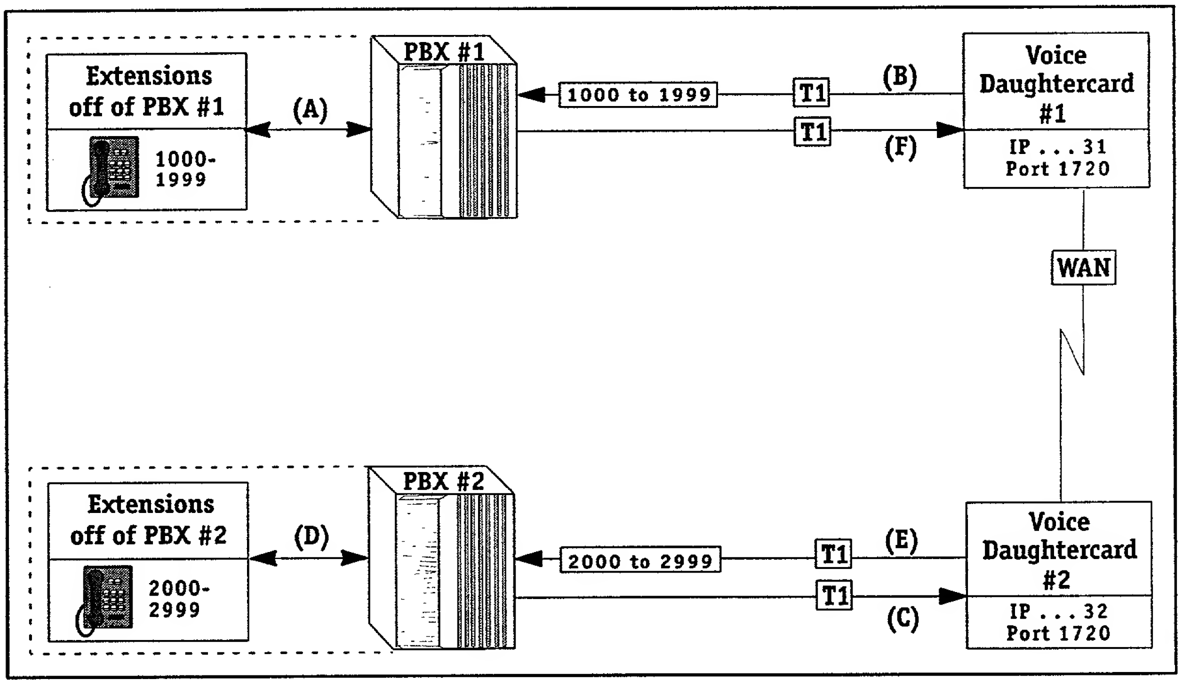
Tbd — Legends to AVNDS examples to be updated according to above trunk list.

Tbd — AVNDS examples (Remarks Section) with features supported and primary CLI commands list incomplete; to be updated.

VoIP Networks without PSTN — Example 1

Four Digit Extensions and Two Voice Daughtercards

This is one of the simplest dialing schemes to implement in a VoIP network. It uses two voice switching daughtercards to translate four digit extensions. Extensions are unique across the entire enterprise network, and the PBX handles all calls to the PSTN. Since incoming and outgoing trunks are separated, this dialing scheme guarantees that no inseize collisions will occur.



Example 1 — Four Digit Extensions and Two Voice Daughtercards

LEGEND for Diagram Components	
PBX #1 Configuration	**PBX #2 Configuration**
— Expects to receive four digits on trunk **(B)** and then uses these digits to route calls to lines **(A)** or trunks **(F)** or **(G)**.	— Expects to receive four digits on trunk **(E)** and then uses these digits to route calls to lines **(D)** or trunks **(C)** or **(H)**.
— Routes calls starting with 1 to lines **(A)**.	— Routes calls starting with 2 to lines **(D)**.
— Routes calls starting with 2 to trunk **(F)**, and then the VoIP network uses these digits to route calls to trunk **(E)**.	— Routes calls starting with 1 to trunk **(C)**, and then the VoIP network uses these digits to route calls to trunk **(G)**.

Remarks

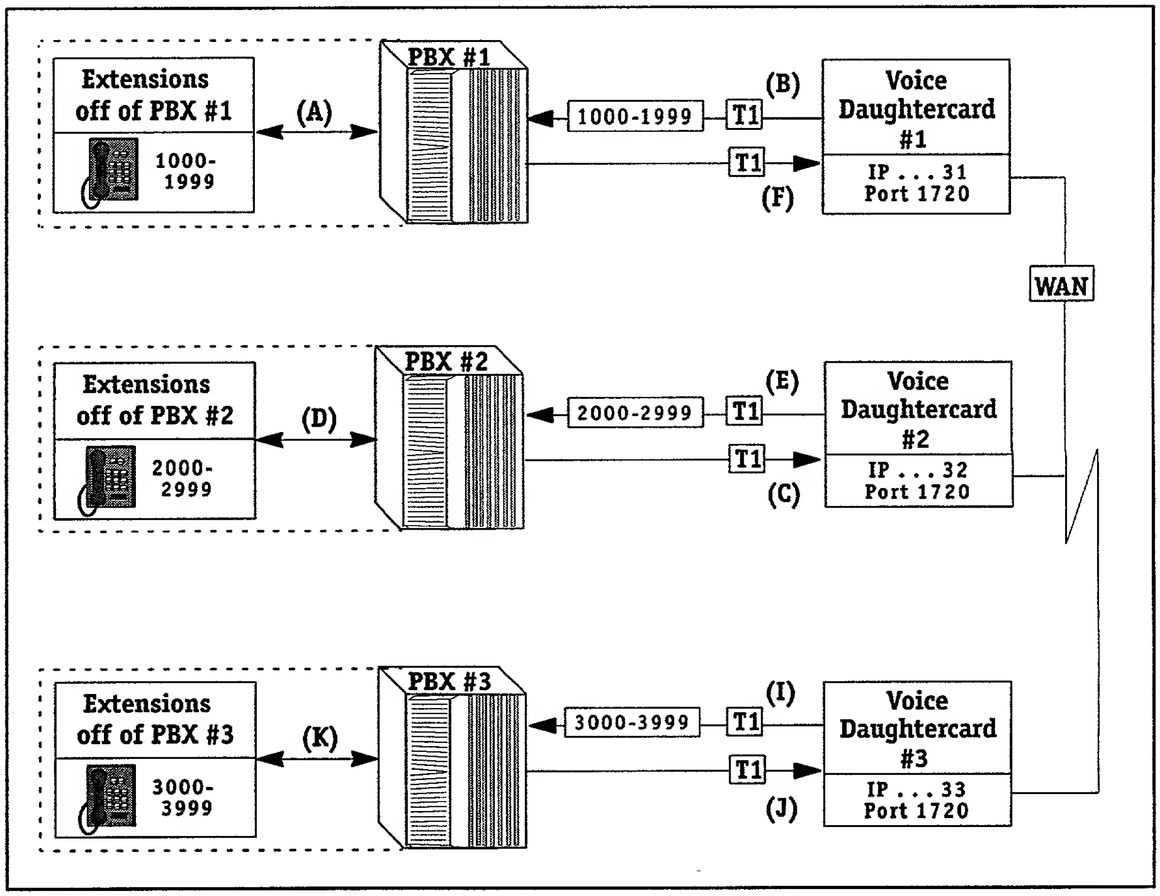
Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard **(A)**	*voice destination local channel* (page 5-244)
Local channel — two hunt groups (24 channels per group/T1) **(F)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Site prefix — no site prefix **(I)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Voice phone group type — four digit local extensions **(L)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Strip digit length — no strip digits **(R)**	*voice phone group strip digit length* (page 5-257)

VoIP Networks without PSTN — Example 2

Trunk Groups and Three Voice Daughtercards

This dialing scheme is used to set up additional sites on an existing VoIP network.



Example 2 — Trunk Groups and Three Voice Daughtercards

LEGEND for Diagram Components		
PBX #1 Configuration	**PBX #2 Configuration**	**PBX #3 Configuration**
— Expects to receive four digits on trunk **(B)** and then uses these digits to route calls to lines **(A)** or trunk **(C)**.	— Expects to receive four digits on **(E)** and then route digits to **(D)**.	— Expects to receive four digits on **(E)** and then route digits to **(D)**.
— Routes VoIP calls to trunk **(F)** and sends four digits to voice daughtercard.	— Routes VoIP calls to trunk **(C)** and sends four digits to voice daughtercard.	— Routes VoIP calls to trunk **(C)** and sends four digits to voice daughtercard.

Remarks

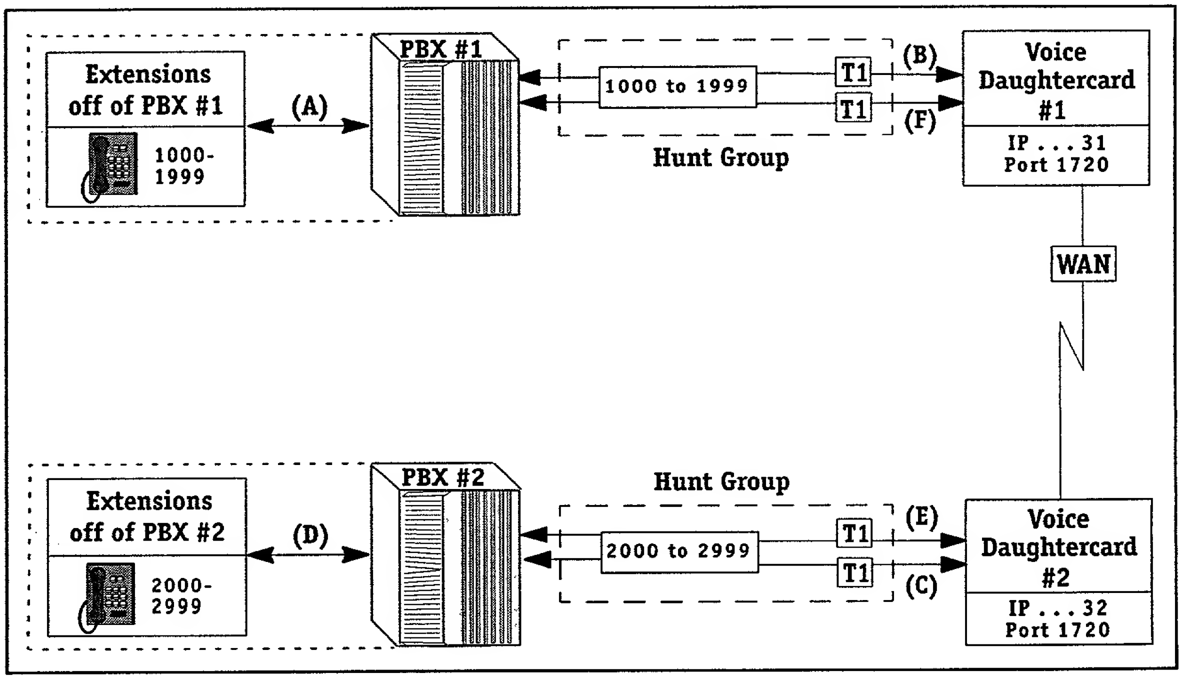
Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard **(A)**	*voice destination local channel* (page 5-244)
Local channel — two hunt groups (24 channels per group/T1) **(F)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Site prefix — no site prefix **(I)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Voice phone group type — four digit local extensions **(L)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Strip digit length — no strip digits **(R)**	*voice phone group strip digit length* (page 5-257)

VoIP Networks without PSTN — Example 3

One Hunt Group (48 Channels Across Two T1s)

This dialing scheme uses one hunt group spanning two T1 lines to make a single 48 channel trunk. Hunt groups relate phone groups and destinations. In the command line syntax, hunt groups are called voice numbering plans.



Example 3 — One Hunt Group (48 Channels Across Two T1s)

LEGEND for Diagram Components	
PBX #1 Configuration	**PBX #2 Configuration**
— Expects to receive four digits on trunk **(B)** and then uses these digits to route calls to lines **(A)** or trunk **(G)**.	— Expects to receive four digits on **(E)** and then uses these digits to route calls to lines **(D)** or trunk **(G)**.
— Routes VoIP calls to trunk **(F)** and sends four digits to voice daughtercard.	— Routes VoIP calls to trunk **(C)** and sends four digits to voice daughtercard.

Remarks

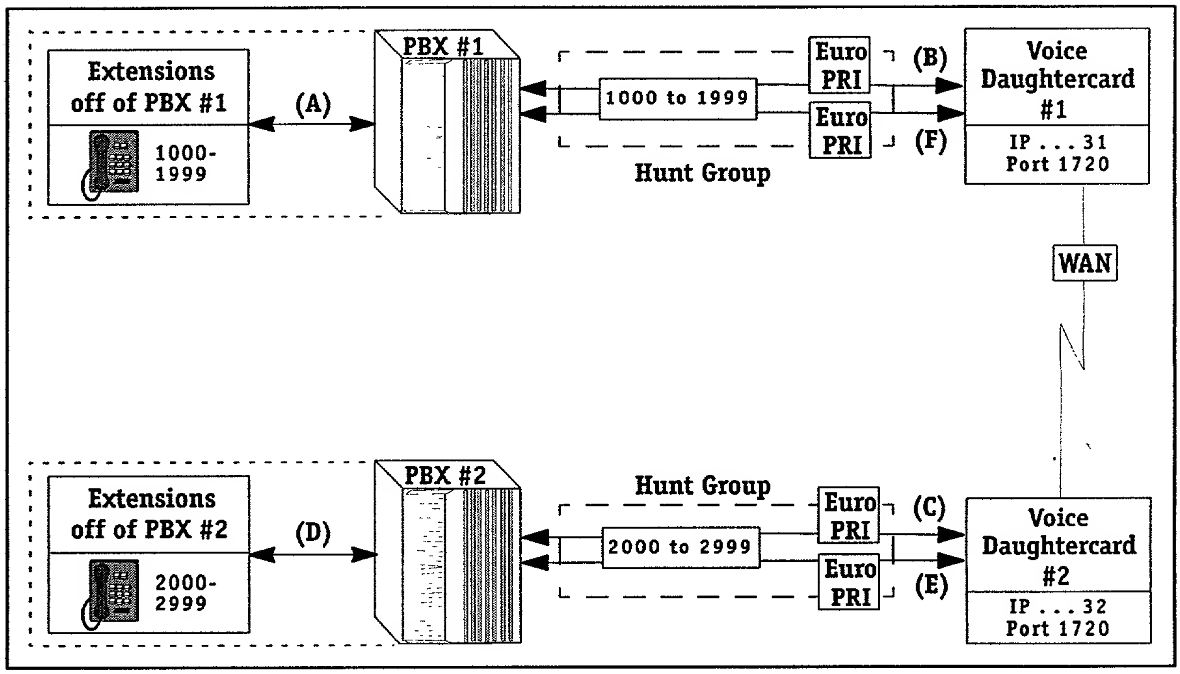
Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard **(A)**	*voice destination local channel* (page 5-244)
Local channel — two hunt groups (24 channels per group/T1) **(F)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Site prefix — no site prefix **(I)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Voice phone group type — four digit local extensions **(L)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Strip digit length — no strip digits **(R)**	*voice phone group strip digit length* (page 5-257)

VoIP Networks without PSTN — Example 4

Two Hunt Groups (60 Channels Across Two E1s)

This dialing scheme uses one hunt group spanning two E1 (Euro PRI) trunks to make a single 60 channel trunk. Hunt groups relate phone groups and destinations. In the command line syntax, hunt groups are called voice numbering plans.



Example 4 — Two Hunt Groups (60 Channels Across Two E1s)

LEGEND for Diagram Components	
PBX #1 Configuration	**PBX #2 Configuration**
— Expects to receive four digits on trunk **(B)** and then uses these digits to route calls to lines **(A)** or trunk **(F)**.	— Expects to receive four digits on trunk **(C)** and then uses these digits to route calls to lines **(D)** or trunk **(E)**.
— Routes VoIP calls to trunk **(F)** and sends four digits to voice daughtercard	— Routes VoIP calls to trunk **(C)** and sends four digits to voice daughtercard.

Remarks

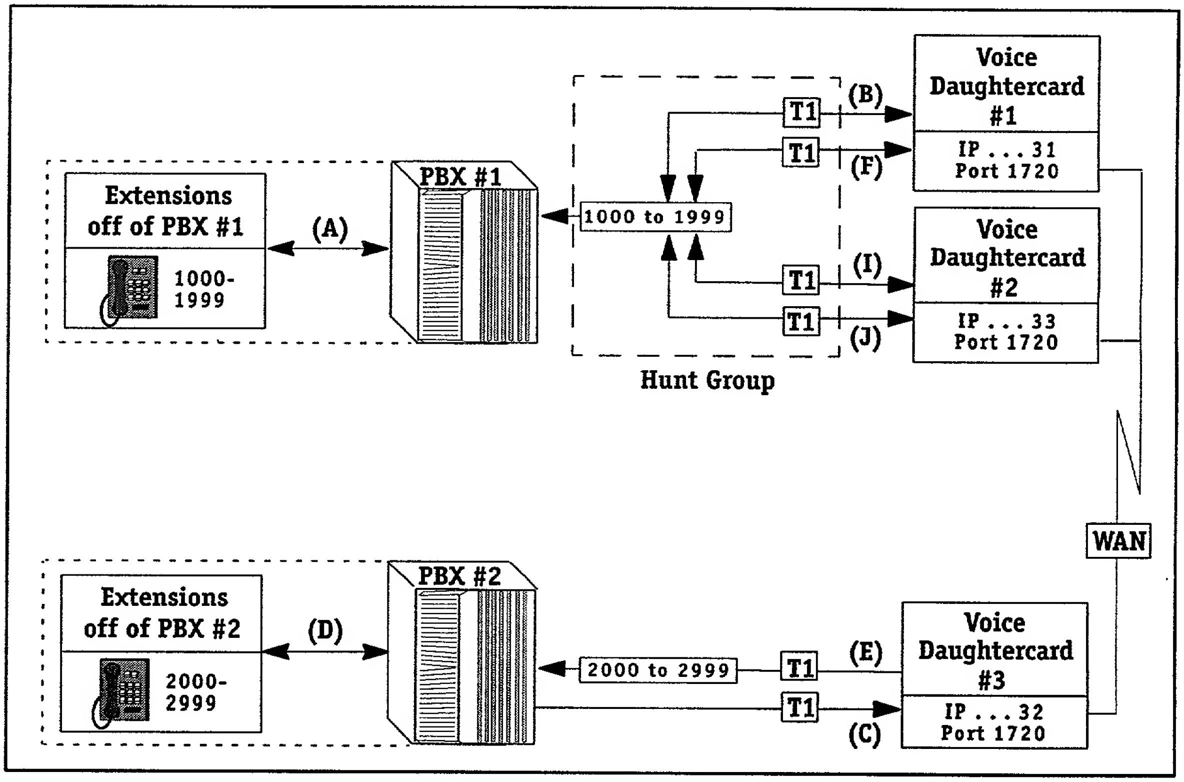
Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard **(A)**	*voice destination local channel* (page 5-244)
Local channel — two hunt groups (60 channels across two E1s) **(H)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Site prefix — no site prefix **(I)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Voice phone group type — four digit local extensions **(L)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Strip digit length — no strip digits **(R)**	*voice phone group strip digit length* (page 5-257)
Digital Interface type — T1 **(W)**	*voice port interface type* (page 5-31)
Digital Interface type — E1 (QSIG) **(X)**	*voice port interface type* (page 5-31)
Digital Interface type — E1 ISDN PRI (Euro PRI) **(Y)**	*voice port interface type* (page 5-31)
Digital Interface type — BRI Euro **(Z)**	*voice port interface type* (page 5-31)

VoIP Networks without PSTN — Example 5

One Hunt Group (96 Channels Across Four T1s)

In this dialing scheme, one hunt group spans four T1 lines using two voice switching daughtercards (spanning two T1 lines each). In this example, the cards are installed in a single VSX motherboard in the same slot of an Omni Switch/Router to provide four T1 lines connected to a PBX. Hunt groups relate phone groups and destinations. In the command line syntax, hunt groups are called voice numbering plans.



Example 5 — One Hunt Group (96 Channels Across Four T1s)

LEGEND for Diagram Components	
PBX #1 Configuration	**PBX #2 Configuration**
— Expects to receive four digits on trunk **(B)** and then uses these digits to route calls to lines **(A)** or trunk **(G)**.	— Expects to receive four digits on **(E)** and then uses these digits to route calls to lines **(D)** or trunk **(H)**.
— Routes VoIP calls to trunk **(F)** and sends four digits to voice daughtercard.	— Routes VoIP calls to trunk **(C)** and sends four digits to voice daughtercard.

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

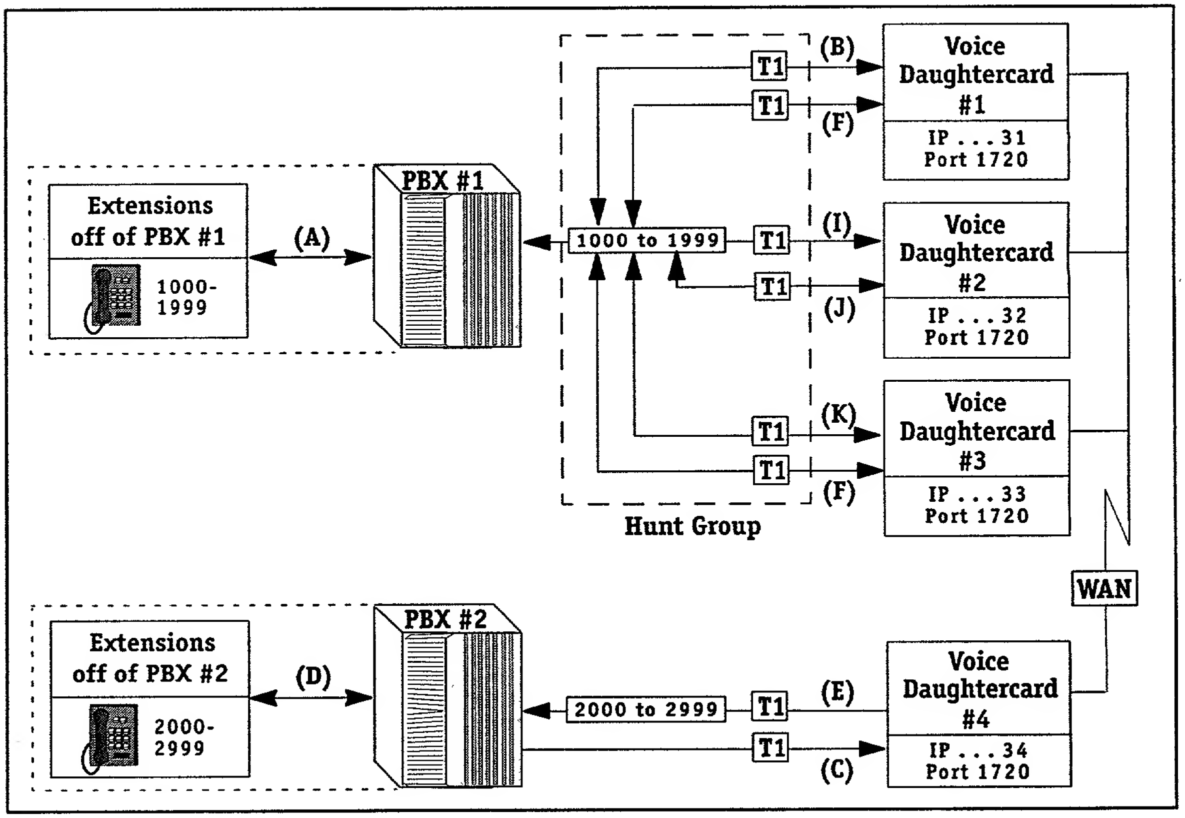
Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard **(A)**	*voice destination local channel* (page 5-244)
Local channel — two hunt groups (24 channels per group/T1) **(F)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Local channel — one hunt group (48 channels across two T1s) **(G)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Site prefix — no site prefix **(I)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Voice phone group type — four digit local extensions **(L)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Strip digit length — no strip digits **(R)**	*voice phone group strip digit length* (page 5-257)

VoIP Networks without PSTN — Example 6

One Hunt Group (144 Channels Across Six T1s)

This dialing scheme has six T1 lines connected to one PBX. In this dialing scheme, one hunt group spans six T1 lines using three voice switching daughtercards (spanning two T1 lines each). In this example, the cards are installed in a two VSX motherboards in the same slot of an Omni Switch/Router to provide six T1 lines connected to a PBX.

Hunt groups relate phone groups and destinations. In the command line syntax, hunt groups are called voice numbering plans.



Example 6 — One Hunt Group (144 Channels Across Six T1s)

LEGEND for Diagram Components	
PBX #1 Configuration	**PBX #2 Configuration**
— Expects to receive four digits on trunk **(B)** and then uses these digits to route calls to lines **(A)** or trunk **(G)**.	— Expects to receive four digits on **(E)** and then uses these digits to route calls to lines **(D)** or trunk **(H)**.
— Routes VoIP calls to trunk **(F)** and sends four digits to voice daughtercard.	— Routes VoIP calls to trunk **(C)** and sends four digits to voice daughtercard.
— Routes all PSTN calls to trunk **(G)**.	— Routes all PSTN calls to trunk **(H)**.

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard **(A)**	*voice destination local channel* (page 5-244)
H.323 gateway to H.323 gatekeeper (RADVision) **(B)**	*voice network h.323 gatekeeper control* (page 5-230) *voice network h.323 gatekeeper mode* (page 5-231) *voice network h.323 gatekeeper address* (page 5-232)
Local channel — one hunt group (48 channels across two T1s) **(G)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Site prefix — no site prefix **(I)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Voice phone group type — four digit local extensions **(L)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Strip digit length — no strip digits **(R)**	*voice phone group strip digit length* (page 5-257)

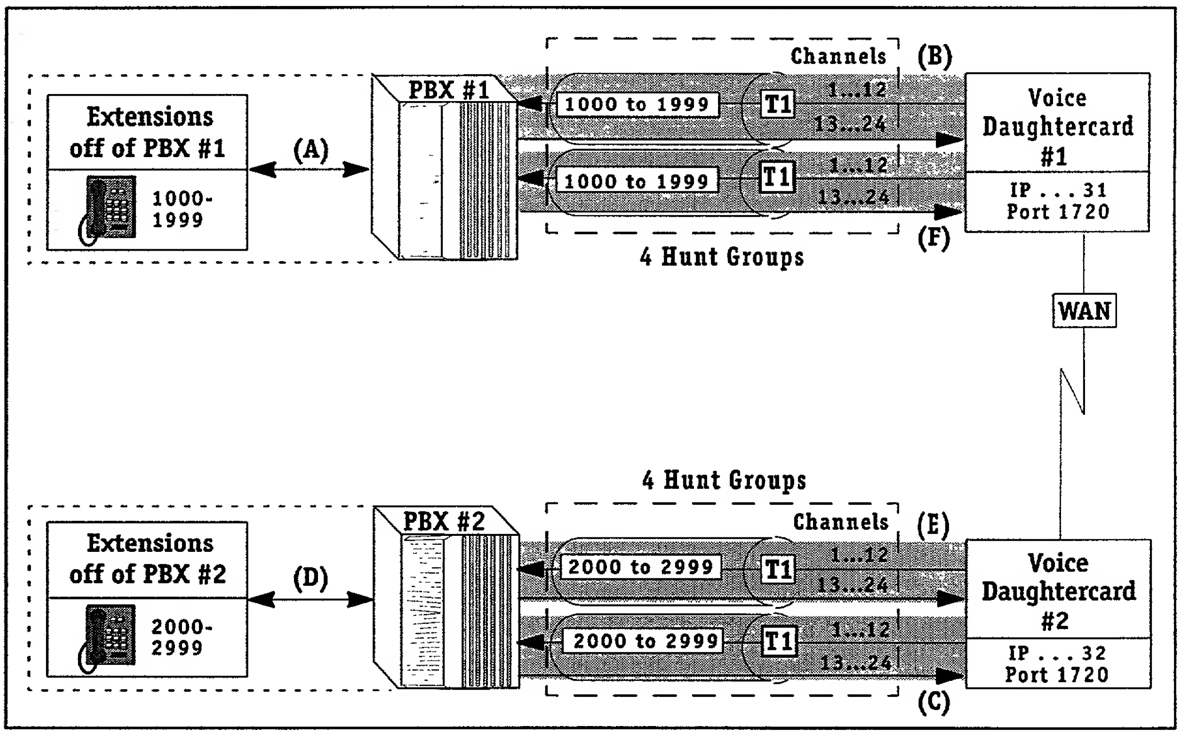
VoIP Networks without PSTN — Example 7

Four Hunt Groups (12 Channels Per Hunt Group)

This dialing scheme is used to split a T1 line in half, and demonstrates one way of having redundant T1 lines on one switch. In this example, one hunt group is half of a T1 line, or 12 channels. Each T1 trunk is split into an incoming and outgoing hunt group.

Dialing Scheme Examples 7 and 8 (showing fractional T1 hunt groups) apply only to digital (VSD) and Euro BRI (VSB) voice switching daughtercards. See Chapter 2, "VoIP Daughtercards" for a description of the various daughtercards, and Chapter 4, "Setup and Installation" for details on installation.

Hunt groups relate phone groups and destinations. In the command line syntax, hunt groups are called voice numbering plans.



Example 7— Four Hunt Groups (12 Channels Per Hunt Group)

LEGEND for Diagram Components	
PBX #1 Configuration	**PBX #2 Configuration**
— Expects to receive four digits on channels 1 through 12 on trunk **(B)** and then uses these digits to route calls to lines **(A)**.	— Expects to receive four digits on **(E)** and then route digits to **(D)**.
— Routes VoIP calls to channels 13 through 24 on trunk **(B)** or channels 13 through 24 on trunk **(F)**.	— Routes VoIP calls to trunk **(K)** or trunk **(L)** and sends four digits to voice daughtercard.

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

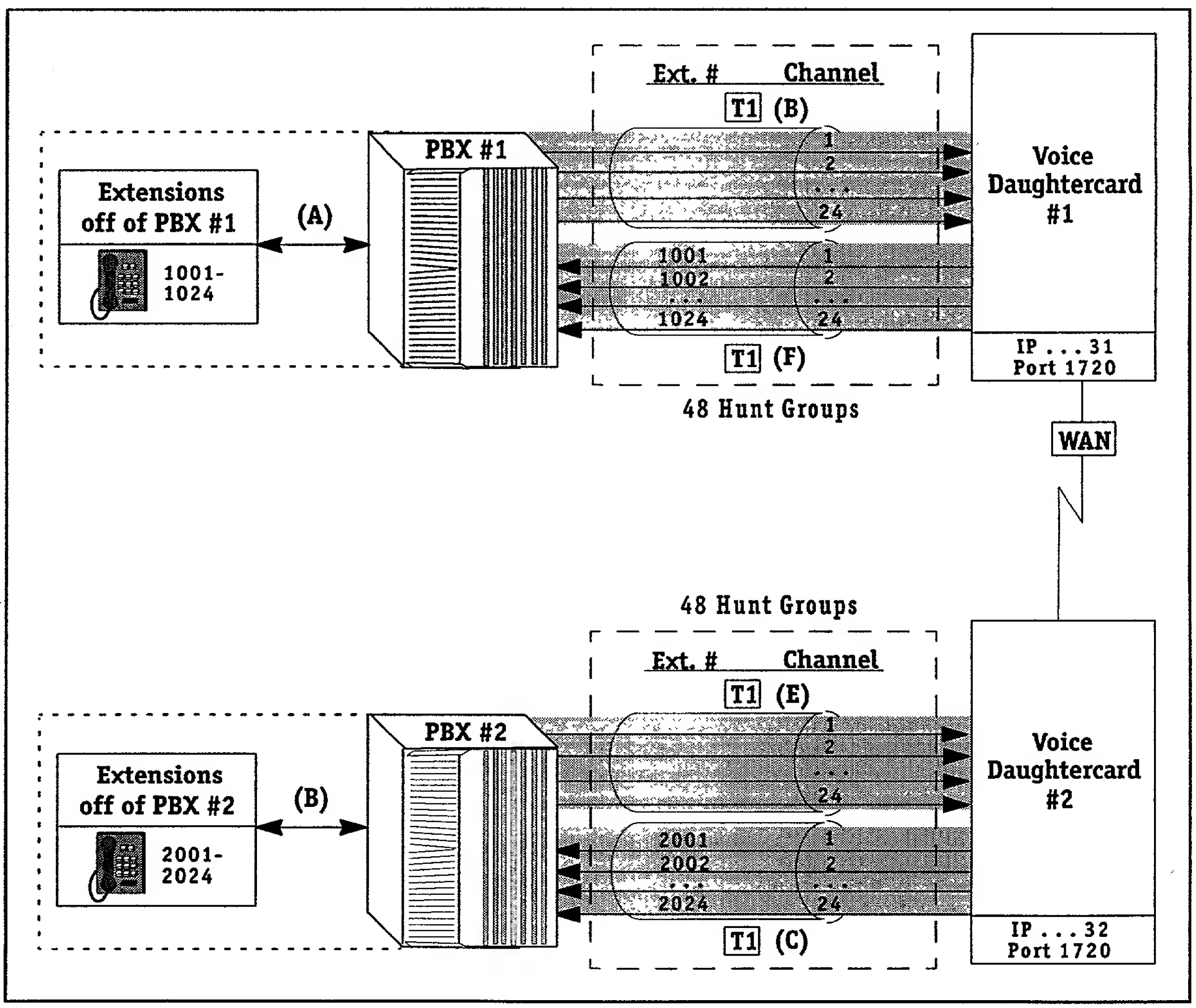
Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard **(A)**	*voice destination local channel* (page 5-244)
Local channel — two hunt groups (24 channels per group/T1) **(F)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Site prefix — no site prefix **(I)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Voice phone group type — four digit local extensions **(L)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Strip digit length — no strip digits **(R)**	*voice phone group strip digit length* (page 5-257)

VoIP Networks without PSTN — Example 8

48 Individual Hunt Groups (One Channel Per Group)

This dialing scheme shows how to divide a single T1 line into smaller (or fractional T1) trunk groups. Additionally, each channel has a unique telephone number and is also associated with a single telephone number. Often this dialing scheme is used to test individual channels on a T1 line, but it can also be used to bypass hunt group behavior. Since each hunt group has only one channel, hunting is, in effect, disabled.

Dialing Scheme Examples 7 and 8 (showing fractional T1 hunt groups) apply only to digital (VSD) and Euro BRI (VSB) voice switching daughtercards. See Chapter 2, "VoIP Daughtercards" for a description of the various daughtercards, and Chapter 4, "Setup and Installation" for details on installation. Hunt groups relate phone groups and destinations. In the command line syntax, hunt groups are called voice numbering plans.



Example 8 — 48 Individual Hunt Groups (One Channel Per Group)

LEGEND for Diagram Components	
PBX #1 Configuration	**PBX #2 Configuration**
— Expects to receive four digits on trunk **(F)** and then uses these digits to route calls to lines **(A)** or trunk **(G)**.	— Expects to receive four digits on trunk **(C)** and then uses these digits to route calls to lines **(B)** or trunk **(G)**.
— Routes VoIP calls to trunk **(B)** and sends four digits to voice daughtercard.	— Routes VoIP calls to trunk **(E)** and sends four digits to voice daughtercard.

Remarks

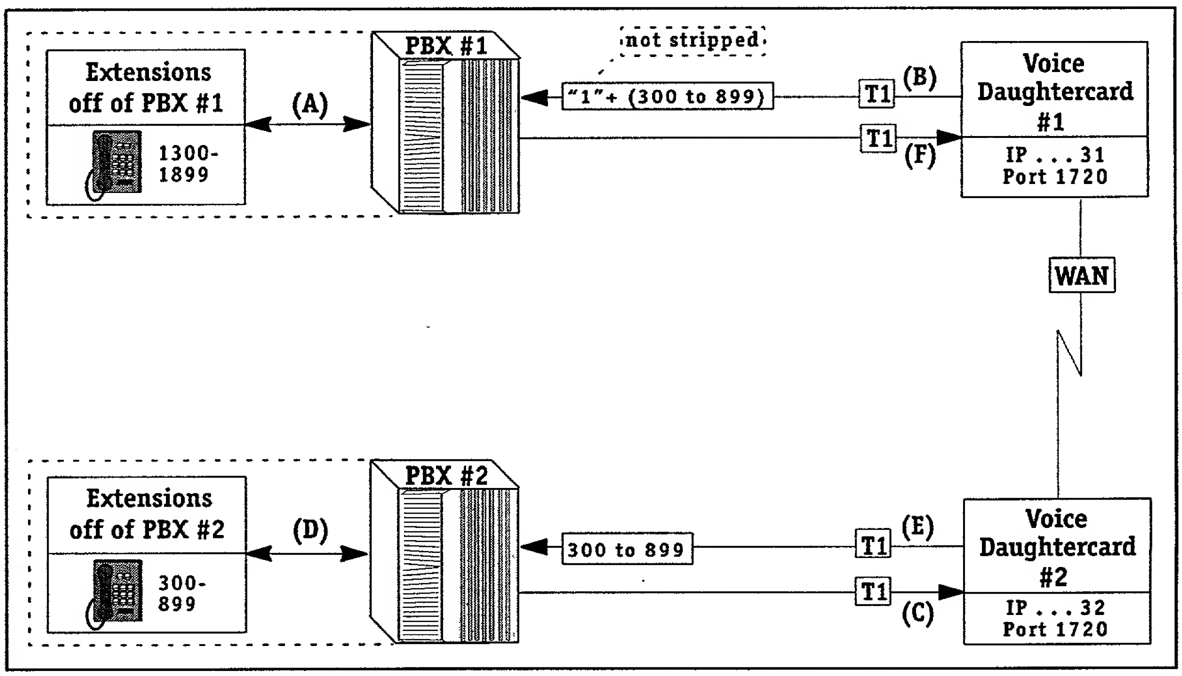
Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard **(A)**	*voice destination local channel* (page 5-244)
Local channel — individual hunt groups (48 channels per group/T1) **(D)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Site prefix — no site prefix **(I)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Voice phone group type — four digit local extensions **(L)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Strip digit length — no strip digits **(R)**	*voice phone group strip digit length* (page 5-257)

VoIP Networks without PSTN — Example 9

Trunk Groups and Mixed Length Extensions

This is another dialing scheme that is relatively simple to implement in a VoIP network, and demonstrates how to mix different extension lengths in one dialing scheme. It uses two voice switching daughtercards to translate a single digit trunk prefix and three digit extensions. The single digit site prefix, rather than the three digits extensions, are unique across the VoIP network. The site prefix digit is used to send the VoIP calls to the correct PBX node. When a caller dials a site prefix, e.g., 1, it routes the call to the corresponding PBX and then dials a prefix to get a specific trunk.



Example 9 — Trunk Groups and Mixed Length Extensions

LEGEND for Diagram Components	
PBX #1 Configuration	**PBX #2 Configuration**
— Expects to receive four digits on trunk **(B)** and then routes digits to lines **(A)**, or trunks **(F)** or **(G)**.	— Expects to receive three digits on trunk **(E)** and then routes digits to lines **(D)**, or trunks **(E)** or **(H)**.
— Routes calls starting with 1 to lines **(A)**.	— Routes calls starting with 3, 4, 5, 6, 7 or 8 to lines **(D)**.
— Routes calls starting with 3, 4, 5, 6, 7 or 8 to trunk **(F)**, then the VoIP network uses the three digits to route calls to trunk **(E)**.	— Routes calls starting with 1 to trunk **(C)**, and then the VoIP network uses all four digits to route calls to trunk **(B)**.
— Routes calls starting with 9 to PSTN.	— Routes calls starting with 9 to PSTN.

Remarks

In the CLI commands, trunk groups are referred to as Site Prefix. Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

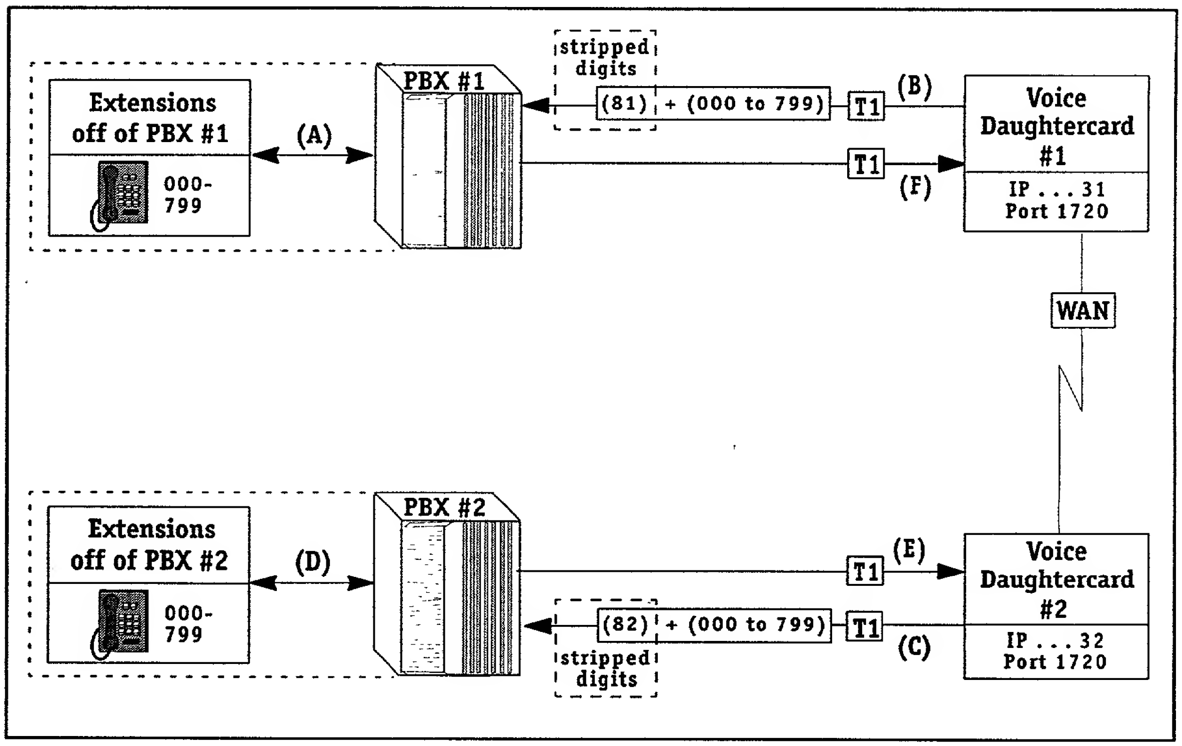
Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard **(A)**	*voice destination local channel* (page 5-244)
Local channel — two hunt groups (24 channels per group/T1) **(F)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Site prefix — single digit **(J)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Site prefix — multiple digits **(J1)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Voice phone group type — three digit local extensions **(K)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Strip digit length — no strip digits **(R)**	*voice phone group strip digit length* (page 5-257)

VoIP Networks without PSTN — Example 10

Strip Digit Length (2)

In this dialing scheme, the PBX uses the first two digits received to route calls. The PBX first dials an "8" to go to the VoIP network. The 2nd digit dialed ("1" or "0") determines the site (PBX) to which the call is sent. A "1" means the call goes to PBX# 1, and a "2" means the call goes to PBX #2. The two-digit prefix is stripped before the voice switching daughtercard sends the digits to the PBX.

Any number used as a site prefix cannot be used for the first digit of any valid extension.



Example 10 — Strip Digit Length (2)

LEGEND for Diagram Components	
PBX #1 Configuration	**PBX #2 Configuration**
— Expects to receive four digits on trunk **(B)** and then uses these digits to route calls to lines **(A)** or trunk **(G)**.	— Expects to receive four digits on **(E)** and then uses these digits to route calls to lines **(D)** or trunk **(G)**.
— Routes VoIP calls to trunk **(F)** and sends four digits to voice daughtercard.	— Routes VoIP calls to trunk **(C)** and sends four digits to voice daughtercard.
— Routes all PSTN calls to trunk **(G)**.	— Routes all PSTN calls to trunk **(H)**.

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard **(A)**	*voice destination local channel* (page 5-244)
Local channel — two hunt groups (24 channels per group/T1) **(F)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Site prefix — no site prefix **(I)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Site prefix — single digit **(J)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Site prefix — multiple digits **(J1)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Voice phone group type — three digit local extensions **(K)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — four digit local extensions **(L)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — eleven digit local extensions **(M)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — NANP extensions **(N)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — PSTN NANP **(P)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Strip digit length — 2 **(T)**	*voice phone group strip digit length* (page 5-257)

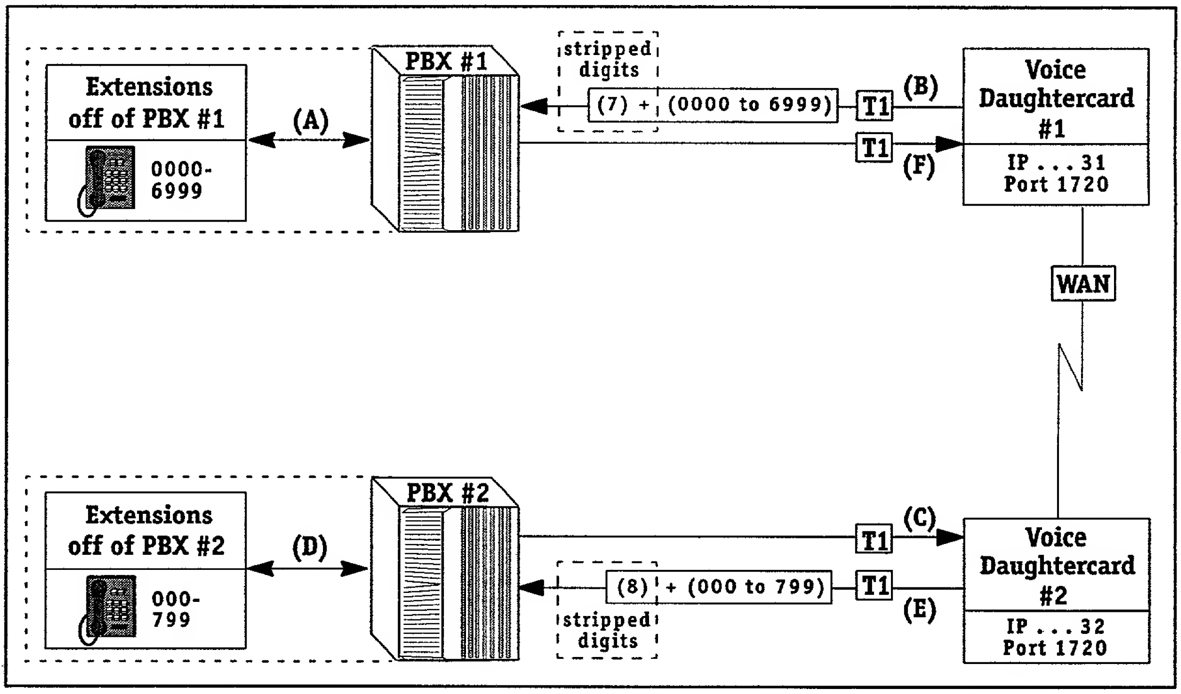
VoIP Networks without PSTN — Example 11

Trunk Groups and Four Digit Extensions

This is one of the more commonly used dialing schemes for VoIP networks. It uses two voice switching daughtercards to translate a single digit trunk prefix and four digit extensions. The single digit site prefix, rather than the four digits extensions, are unique across the VoIP network. This enables each PBX site to handle the same or overlapping phone extensions.

The site prefix digit is used to send the VoIP calls to the correct PBX node. When a caller dials a specific site prefix, e.g., 7, it routes the call to the corresponding PBX and then dials a prefix to get a specific trunk.

Any number used as a site prefix cannot be used for the first digit of any valid extension.



Example 11— Trunk Groups and Four Digit Extensions

LEGEND for Diagram Components	
PBX #1 Configuration	**PBX #2 Configuration**
— Expects to receive four digits on trunk **(B)** and then uses these digits to route calls to lines **(A)** or trunk **(F)**.	— Expects to receive four digits on trunk **(E)** and then uses these digits to route calls to lines **(D)** or trunk **(C)**.
— Routes calls starting with 0, 1, 2, 3, 4, 5 and 6 to lines **(A)**.	— Routes calls starting with 0, 1, 2, 3, 4, 5 and 6 to lines **(D)**.
— Routes calls starting with 8 to trunk **(F)**, then the VoIP network strips the prefix digit of 8 and forwards the remaining four digits to trunk **(E)**	— Routes calls starting with 7 to trunk **(E)**, then the VoIP network strips the prefix digit of 8 and forwards the remaining four digits to trunk **(F)**.
— Routes calls starting with 9 to PSTN trunk **(G)**.	— Routes calls starting with 9 to PSTN trunk **(H)**.

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard **(A)**	*voice destination local channel* (page 5-244)
Local channel — two hunt groups (24 channels per group/T1) **(F)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Site prefix — single digit **(J)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Voice phone group type — four digit local extensions **(L)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Strip digit length — 1 **(S)**	*voice phone group strip digit length* (page 5-257)

VoIP Networks without PSTN — Example TBD

One Trunk Group and Eleven Digit Extensions

This dialing scheme can be used with two voice daughtercards to translate seven digit trunk prefixes and four digit extensions. The seven digit site prefix, rather than the four digits extensions, are unique across the VoIP network. This enables each PBX site to handle the same or overlapping phone extensions.

The site prefix digits are used to send the VoIP calls to the correct PBX node. When a caller dials a specific site prefix, e.g., 1-603-598, it routes the call to the corresponding PBX and then dials a prefix to get a specific trunk.

Any number used as a site prefix cannot be used for the first digits of any valid extension.

Example TBD — One Trunk Group and Eleven Digit Extensions

LEGEND for Diagram Components	
PBX #1 Configuration	**PBX #2 Configuration**
— Expects to receive four digits on trunk **(B)** and then uses these digits to route calls to lines **(D)**.	— Expects to receive four digits on trunk **(E)** and then uses these digits to route calls to lines **(A)**.
— Routes all outbound calls with 2000-2999 extensions to lines **(A)**.	— Routes all outbound calls with 2000-2999 extensions to lines **(D)**.
— Routes all 0 . . ., 411, 911 and 1-npa . . . calls to trunk **(B)**, and then the VoIP network uses these digits to route calls to trunks **(G)** or **(E)**.	— Routes all 0 . . ., 411, 911 and 1-npa . . calls to trunk **(E)**, and then the VoIP network uses these digits to route calls to trunks **(H)** or **(B)**.

Remarks

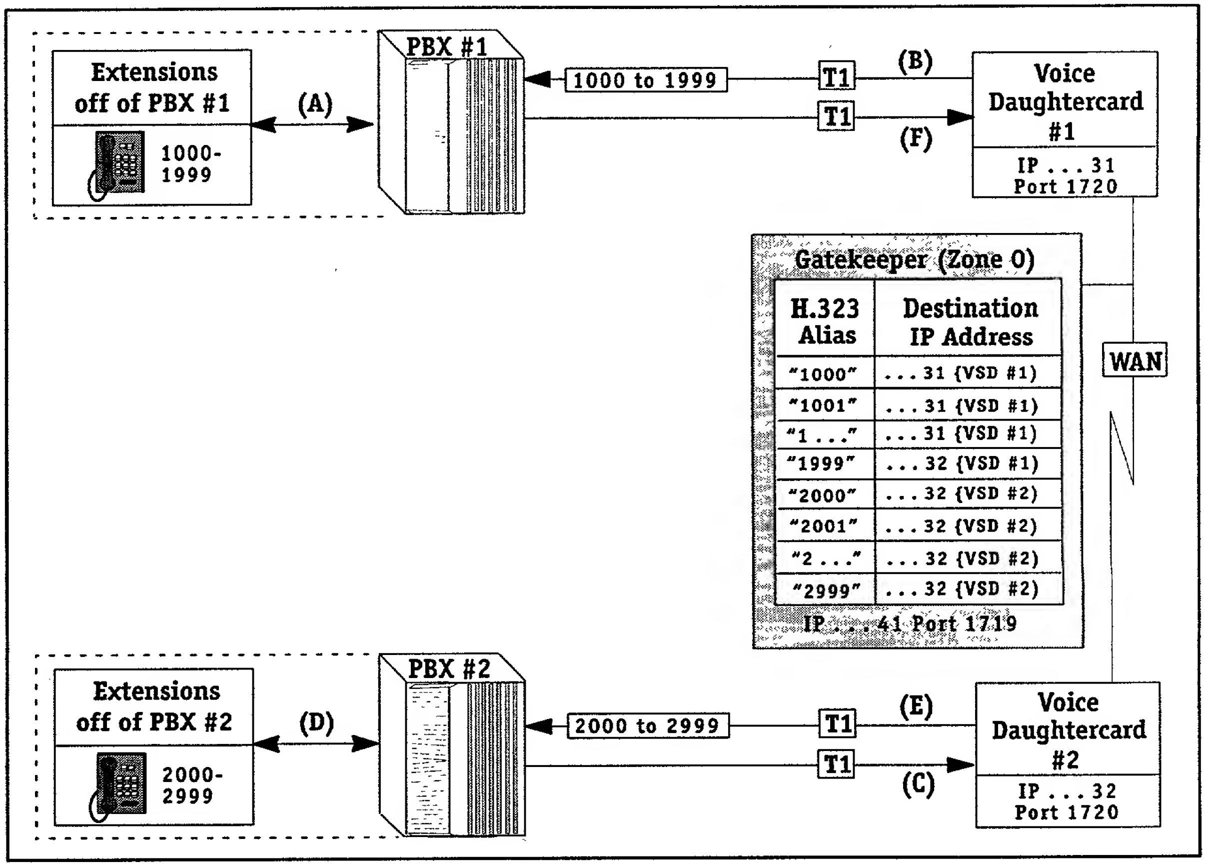
Supported VoIP features and main CLI commands used with this dialing scheme are as follows, and are applicable to the outbound, inbound DID and North American VoIP calls.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard **(A)**	*voice destination local channel* (page 5-244)
Local channel — two hunt groups (24 channels per group/T1) **(F)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Site prefix — multiple digits **(J1)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Voice phone group type — NANP extensions **(N)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — PSTN NANP **(P)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — PSTN INTL **(Q)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Strip digit length — 7 **(V)**	*voice phone group strip digit length* (page 5-257)

VoIP Networks without PSTN — Example 12

H.323 Gatekeeper

This dialing scheme is used to connect voice switching daughtercards to an external H.323 RADVision (or other third-party) gatekeeper.



Example 12 — H.323 Gatekeeper

LEGEND for Diagram Components	
PBX #1 Configuration	**PBX #2 Configuration**
— Expects to receive four digits on trunk **(F)** and then uses these digits to route calls to lines **(A)** or trunk **(F)**.	— Expects to receive four digits on trunk **(C)** and then uses these digits to route calls to lines **(D)** or trunk **(E)**.
— Routes VoIP calls with x2000 to x2999 extension to **(F)** and sends all four digits to voice daughtercard.	— Routes VoIP calls with x1000 to x1999 extensions to **(C)** and sends all four digits to voice daughtercard.
— Routes all PSTN calls to trunk **(G)**.	— Routes all PSTN calls to trunk **(H)**.

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard **(A)**	*voice destination local channel* (page 5-244)
H.323 gateway to H.323 gatekeeper (RADVision) **(B)**	*voice network h.323 gatekeeper control* (page 5-230) *voice network h.323 gatekeeper mode* (page 5-231) *voice network h.323 gatekeeper address* (page 5-232)
Local channel — two hunt groups (24 channels per group/T1) **(F)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Site prefix — no site prefix **(I)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Voice phone group type — four digit local extensions **(L)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Strip digit length — no strip digits **(R)**	*voice phone group strip digit length* (page 5-257)

VoIP Networks with PSTN — Example 13

North American PSTN and VoIP Calls

This dialing scheme is used for calls going through the North American PSTN and the VoIP Network. The following four diagrams are used to demonstrate how these calls are handled:

- North American PSTN Calls — Overview
- North American PSTN Calls — Outbound
- North American PSTN Calls — Inbound
- North American PSTN Calls — VoIP Network

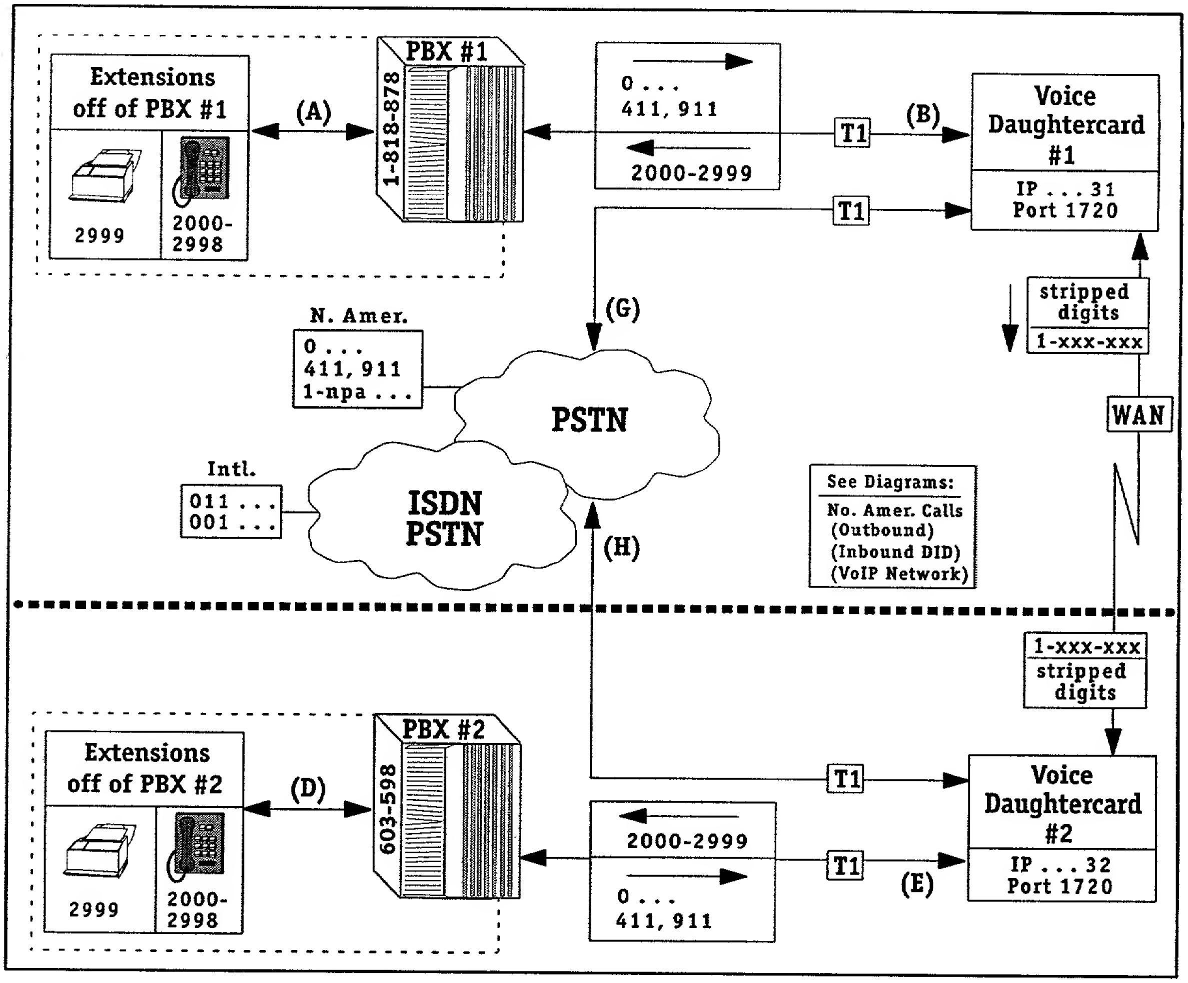
Note that DID is used only on inbound calls, and that all calls are connected using the North American Numbering Plan (NANP) that also includes Canada. See Dialing Scheme Example 15 for details on North American and International calls going through the PSTN or the VoIP network.

Voice daughtercards using either North American or International PSTN dialing schemes cannot handle PBXs with extensions starting with 0, 1, 411 or 911. The following extensions are also not allowed: 0000 to 0999, 1000 to 1999, 4110 to 4119, and 9110 to 9119.

North American PSTN Calls — Overview

In this dialing scheme two voice switching daughtercards are used to translate area codes and telephone numbers. All eleven digits in the telephone numbers are unique across the VoIP network, and the voice switching daughtercards are responsible for all telephone number routing. 411 and 911 calls can be handled as well using this dialing scheme, also referred to as "Drop and Insert." Minimal to no PBX re-configuration is required; however, due to less than 99.995% reliability of Voice over IP networks, this dialing scheme is not recommended unless "passthrough" is used on some channels.

To call a 2000 extension off of PBX#1, the caller dials an eleven digit NANP telephone number. In the overview diagram below, the voice daughtercard strips off the first seven digits, and then forwards the last four digits of the dialed number.



Example 13 — North American PSTN Calls (Overview)

LEGEND for Diagram Components	
PBX #1 Configuration	**PBX #2 Configuration**
— Expects to receive four digits on trunk **(B)** and then uses these digits to route calls to lines **(D)**.	— Expects to receive four digits on trunk **(E)** and then uses these digits to route calls to lines **(A)**.
— Routes all outbound calls with 2000-2999 extensions to lines **(A)**.	— Routes all outbound calls with 2000-2999 extensions to lines **(D)**.
— Routes all 0 . . ., 411, 911 and 1-npa . . . calls to trunk **(B)**, and then the VoIP network uses these digits to route calls to trunks **(G)** or **(E)**.	— Routes all 0 . . ., 411, 911 and 1-npa . . . calls to trunk **(E)**, and then the VoIP network uses these digits to route calls to trunks **(H)** or **(B)**.

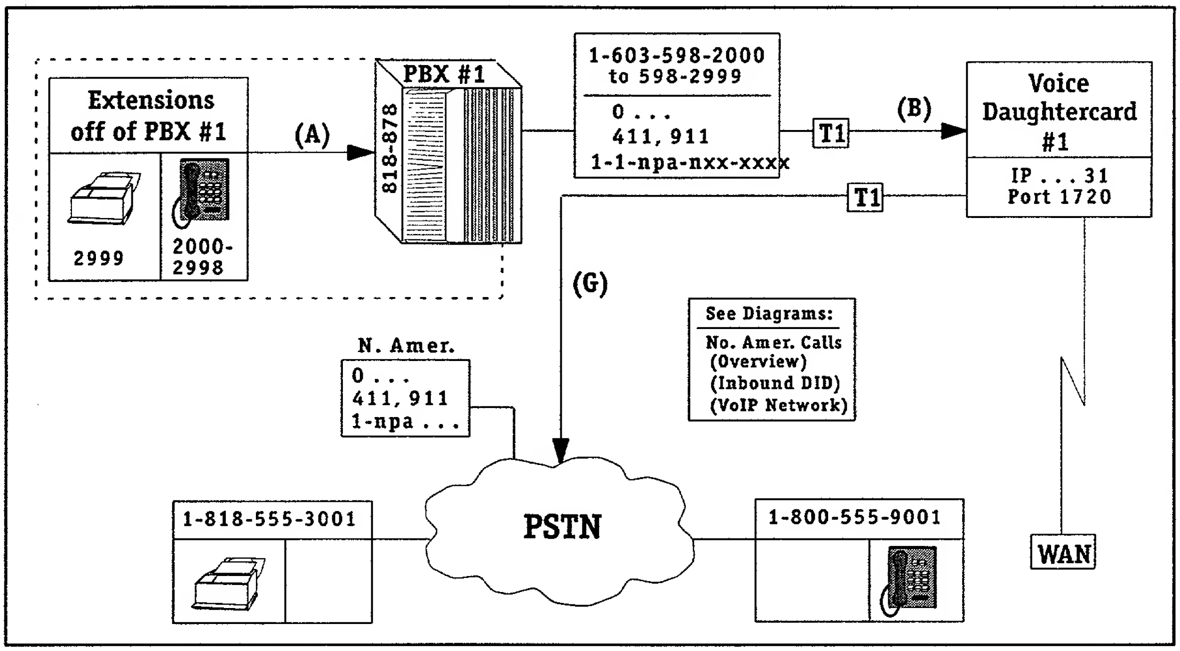
Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows, and are applicable to the outbound, inbound DID and North American VoIP calls.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard **(A)**	*voice destination local channel* (page 5-244)
Local channel — two hunt groups (24 channels per group/T1) **(F)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Site prefix — multiple digits **(J1)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Voice phone group type — NANP extensions **(N)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — PSTN NANP **(P)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — PSTN INTL **(Q)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Strip digit length — 7 **(V)**	*voice phone group strip digit length* (page 5-257)

North American PSTN Calls — Outbound

This diagram demonstrates how outbound North American PSTN calls are sent to the PSTN.

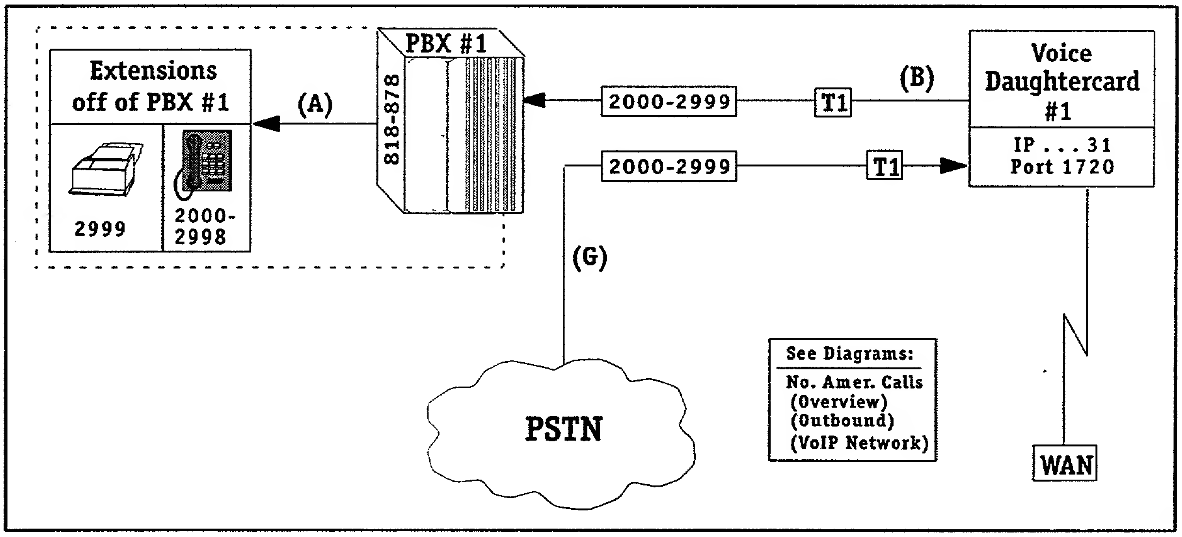


Example 13 — North American PSTN Calls (Outbound)

LEGEND for Diagram Components	
PBX #1 Configuration	**Voice Daughtercard #1 Configuration**
— Routes all calls to trunk **(B)** and sends four digits to voice daughtercard. See next diagram (No. Amer. PSTN Inbound) trunk **(B)** information.	— Expects to receive one to 24 digits on trunk **(B)**, and then uses these digits to route calls to either trunk **(G)** or the VoIP network.
	— Strips the first seven digits of all 1-603-598-2000 to 2999 calls, and then routes the remaining four digits to the VoIP network.
	— Routes all 0 . . . , 411, 911 or 1-npa . . . calls to trunk **(G)**.

North American PSTN Calls — Inbound DID (Direct Inward Dial)

This diagram demonstrates how inbound North American DID calls from the PSTN are handled.



Example 1'3 — North American PSTN Calls (Inbound DID)

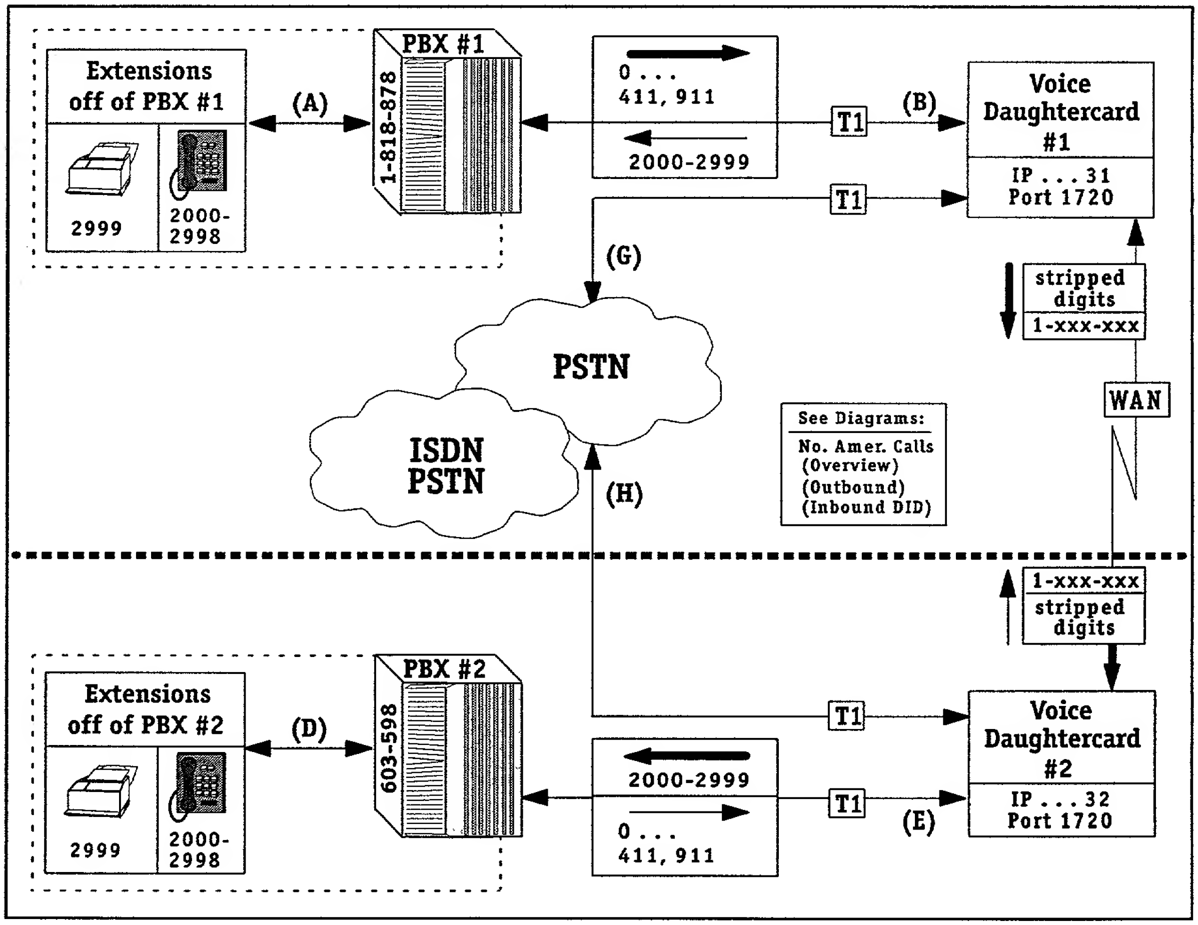
LEGEND for Diagram Components	
PBX #1 Configuration	**Voice Daughtercard #1 Configuration**
See next diagram (No. Amer. PSTN Inbound) trunk **(B)** information.	— Expects to receive four digits on trunk **(G)**, and then uses these digits to route calls to either trunk **(B)**.

North American PSTN Calls — VoIP Network

This diagram demonstrates how North American calls are handled in a VoIP network.

When an 818-878-2000 extension is called from a 603-589-2000 extension, the PBX routes calls to trunk B and then sends 1-603-598-2000 number to Voice Daughtercard #1.

The Voice Daughtercard #1 strips the 1st seven digits and forwards the last four digits across the WAN to Voice Daughtercard #2. These four digits are then forwarded to Trunk E. PBX #2 receives the four digits and then routes the call to the appropriate extension.



Example 13 — North American PSTN Calls (VoIP Network)

LEGEND for Diagram Components	
Voice Daughtercard #1 Configuration	**Voice Daughtercard #2 Configuration**
— Expects to receive four digits on trunk **(B)** and then uses these digits to route calls to lines **(A)** or trunk **(G)**.	— Expects to receive four digits on **(E)** and then uses these digits to route calls to lines **(D)** or trunk **(H)**.
— Routes VoIP calls to trunk **(F)** and sends four digits to voice daughtercard.	— Routes VoIP calls to trunk **(C)** and sends four digits to voice daughtercard.
— Routes all 1-npa-nxx-xxxx PSTN calls to trunk **(G)**.	— Routes all 1-npa-nxx-xxxx PSTN calls to trunk **(H)**.
— Routes all PSTN calls to trunk **(G)**.	— Routes all PSTN calls to trunk **(H)**.

VoIP Networks with PSTN — Dialing Scheme Example 14

North American PSTN, International PSTN and VoIP Calls

This dialing scheme is used for calls going through the North American PSTN, International PSTN and the VoIP network. The following four diagrams are used to demonstrate how these calls are handled:

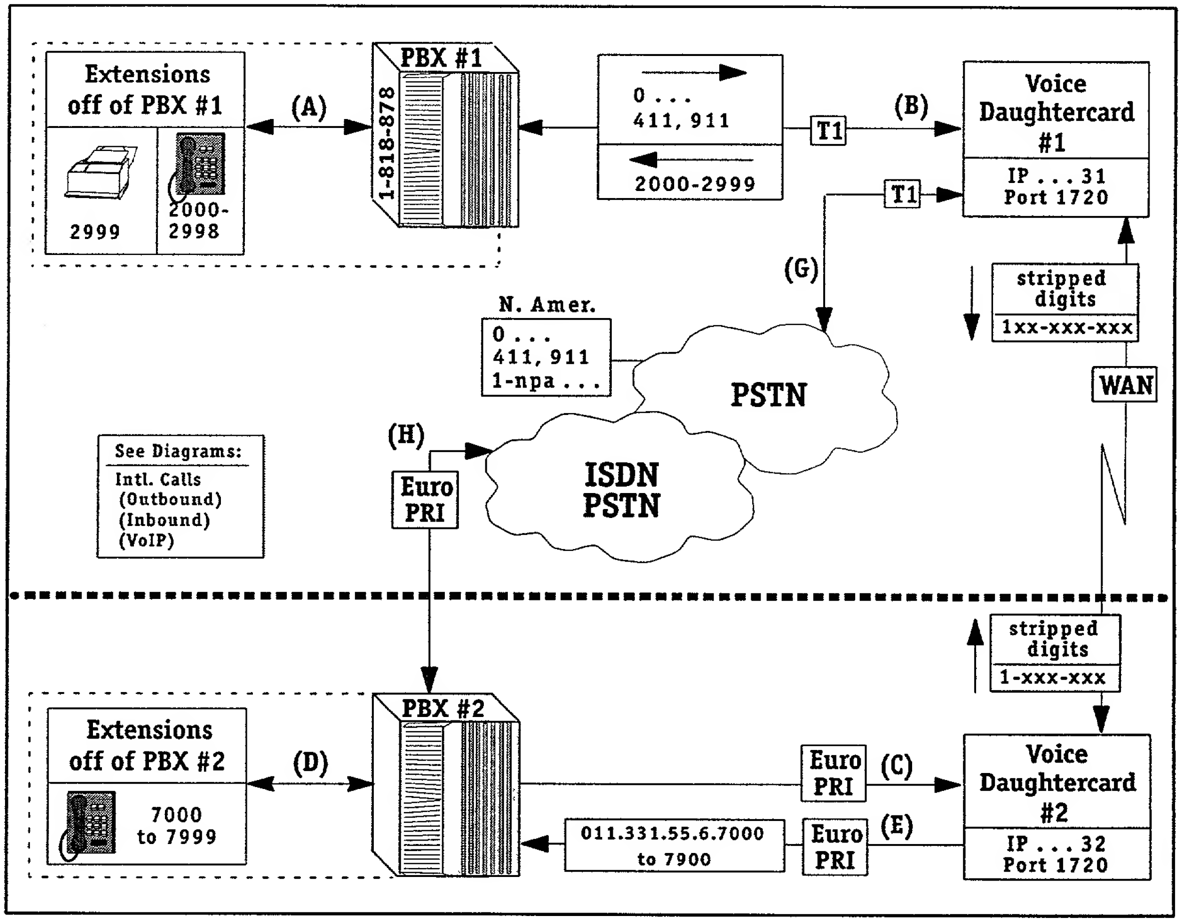
- North American PSTN, International PSTN and VoIP Calls — Overview (*Not available this release*)
- North American PSTN and International PSTN Calls — Outbound (*Not available this release*)
- North American PSTN and International PSTN Calls — Inbound (*Not available this release*)
- North American PSTN and International PSTN Calls — VoIP Network (*Not available this release*)

See Dialing Scheme Example 13 for details on calls going only through the North American PSTN and the VoIP network.

Voice daughtercards using either North American or International PSTN dialing schemes cannot handle PBXs with extensions starting with 0, 1, 411 or 911. The following extensions are also not allowed: 0000 to 0999, 1000 to 1999, 4110 to 4119, and 9110 to 9119.

International PSTN Calls — Overview

This dialing scheme is used to handle international telephone calls. *Not available this release.*



Example 14 — International PSTN Calls (Overview)

LEGEND for Diagram Components	
PBX #1 Configuration	**PBX #2 Configuration**
— Expects to receive four digits on trunk **(B)** and then uses these digits to route calls to lines **(A)** or trunk **(G)**.	— Expects to receive four digits on trunk **(E)** and then route digits to to lines **(D)** or trunk **(H)**.
— Trunk **(B)** is an incoming only trunk. The PBX never outseizes on this trunk.	— Trunk **(E)** is an incoming only trunk. The PBX never outseizes on this trunk.
— Trunk **(F)** is an outgoing only trunk. The PBX never inseizes on this trunk.	— Trunk **(C)** is an outgoing only trunk. The PBX never inseizes on this trunk.
— Routes VoIP calls to trunk **(F)** and sends four digits to voice daughtercard.	— Routes VoIP calls to trunk **(C)** and sends four digits to voice daughtercard.
— Routes all 1-npa-nxx-xxxx PSTN calls to trunk **(G)**.	— Routes all 1-npa-nxx-xxxx PSTN calls to trunk **(H)**.
— Routes all PSTN calls to trunk **(G)**.	— Routes all PSTN calls to trunk **(H)**.

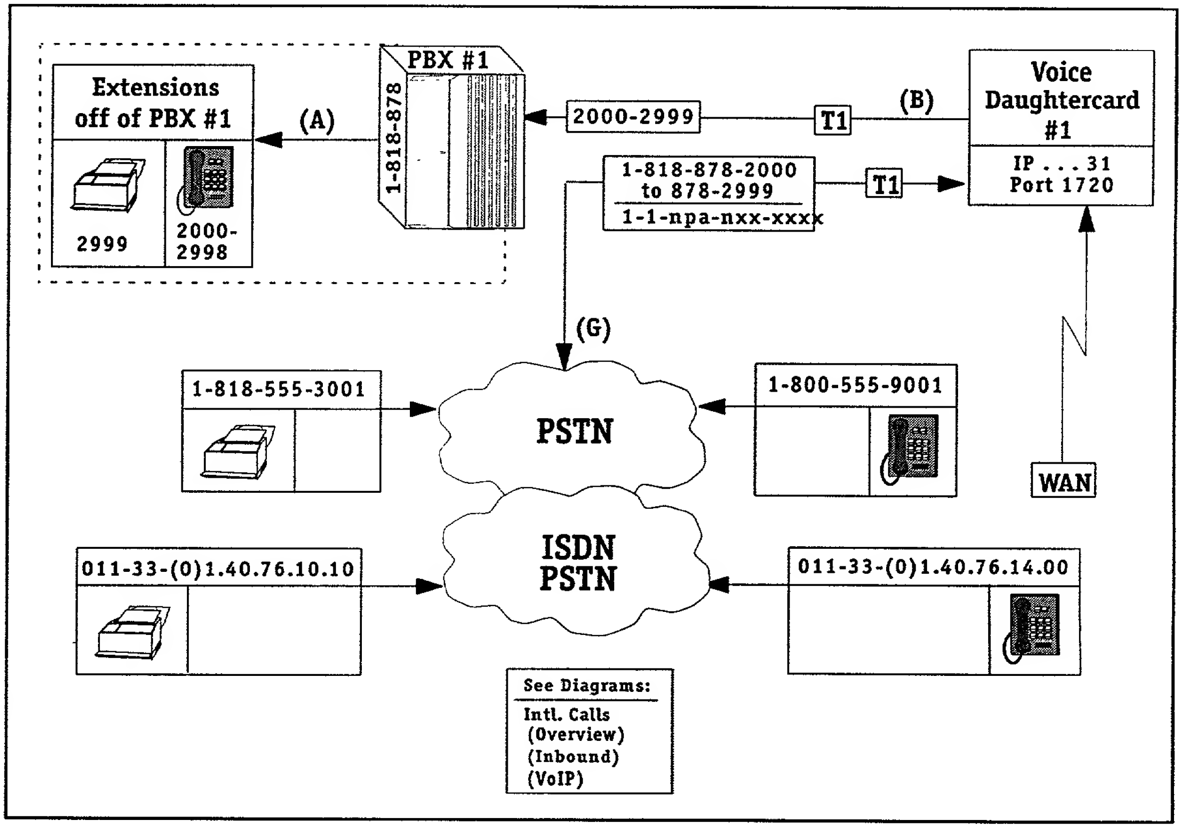
Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows, and are applicable to the outbound, inbound and VoIP PSTN International calls.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard **(A)**	*voice destination local channel* (page 5-244)
Local channel — two hunt groups (24 channels per group/T1) **(F)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Site prefix — no site prefix **(I)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Site prefix — multiple digits **(J1)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Voice phone group type — four digit local extensions **(L)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — NANP extensions **(N)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — INTL extension **(O)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — PSTN NANP **(P)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — PSTN INTL **(Q)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Strip digit length — no strip digits **(R)**	*voice phone group strip digit length* (page 5-257)
Strip digit length — 7 **(V)**	*voice phone group strip digit length* (page 5-257)

International PSTN Calls — Outbound

This dialing scheme is used to handle outbound international telephone calls. *Not available this release.*

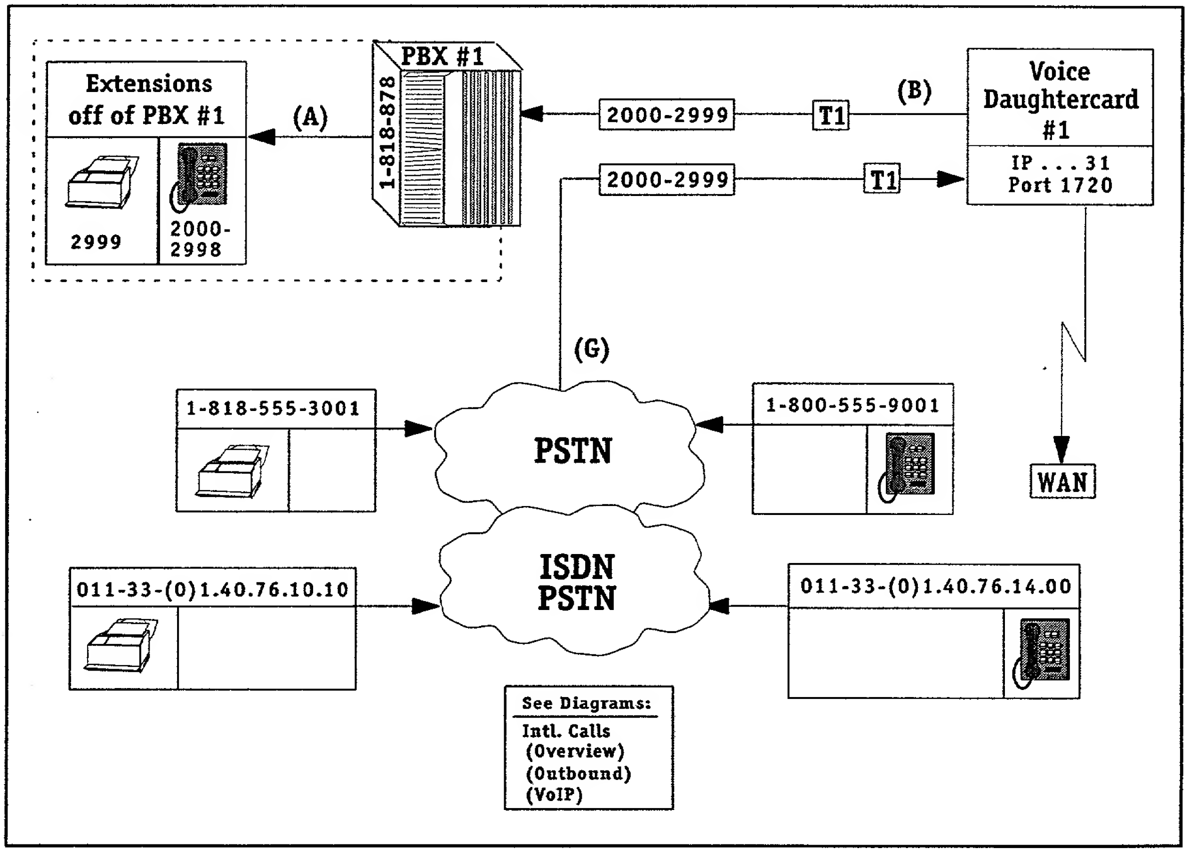


Example 14 — International PSTN Calls (Outbound)

LEGEND for Diagram Components	
PBX #1 Configuration	**PBX #2 Configuration**
— Expects to receive four digits on trunk **(B)** and then uses these digits to route calls to lines **(A)** or trunk **(G)**.	— Expects to receive four digits on **(E)** and then uses these digits to route calls to lines **(D)** or trunk **(H)**.
— Routes VoIP calls to trunk **(F)** and sends four digits to voice daughtercard.	— Routes VoIP calls to trunk **(C)** and sends four digits to voice daughtercard.
— Routes all 1-1-npa-nxx-xxxx PSTN calls to trunk **(G)**.	— Routes all 1-1-npa-nxx-xxxx PSTN calls to trunk **(H)**.
— Routes all PSTN calls to trunk **(G)**	— Routes all PSTN calls to trunk **(H)**.

International PSTN Calls — Inbound

This dialing scheme is used to handle inbound international telephone calls. *Not available this release.*

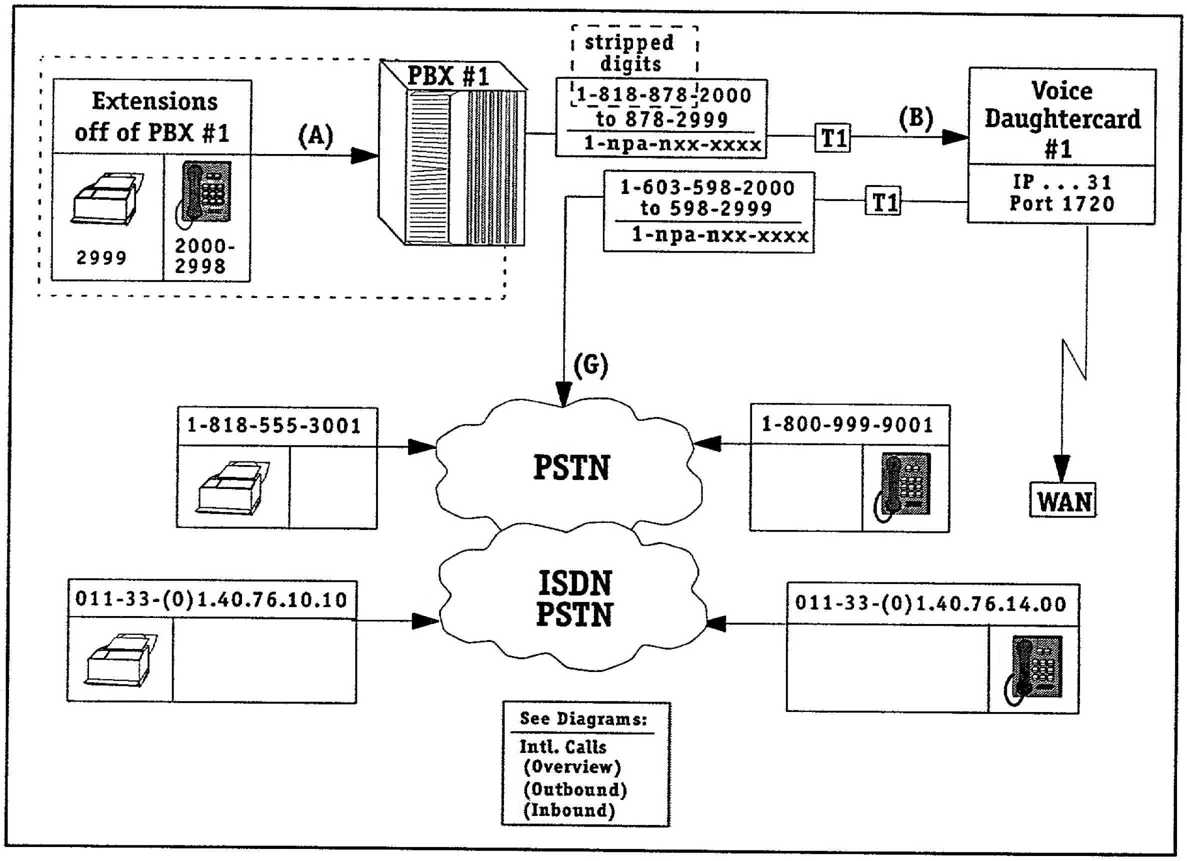


Example 14 —International PSTN Calls (Inbound)

LEGEND for Diagram Components	
PBX #1 Configuration	**PBX #2 Configuration**
— Expects to receive four digits on trunk **(B)** and then uses these digits to route calls to lines **(A)** or trunk **(G)**.	— Expects to receive four digits on trunk **(E)** and then uses these digits to route calls to lines **(D)** or trunk **(H)**.
— Routes VoIP calls to trunk **(F)** and sends four digits to voice daughtercard.	— Routes VoIP calls to trunk **(C)** and sends four digits to voice daughtercard.
— Routes all 1-1-npa-nxx-xxxx PSTN calls to trunk **(G)**.	— Routes all 1-1-npa-nxx-xxxx PSTN calls to trunk **(H)**.
— Routes all PSTN calls to trunk **(G)**.	— Routes all PSTN calls to trunk **(H)**.

International PSTN Calls — VoIP Network

This diagram demonstrates what happens on international calls handled between voice daughtercards. *Not available this release.*



Example 14 — PSTN International Calls (VoIP Network)

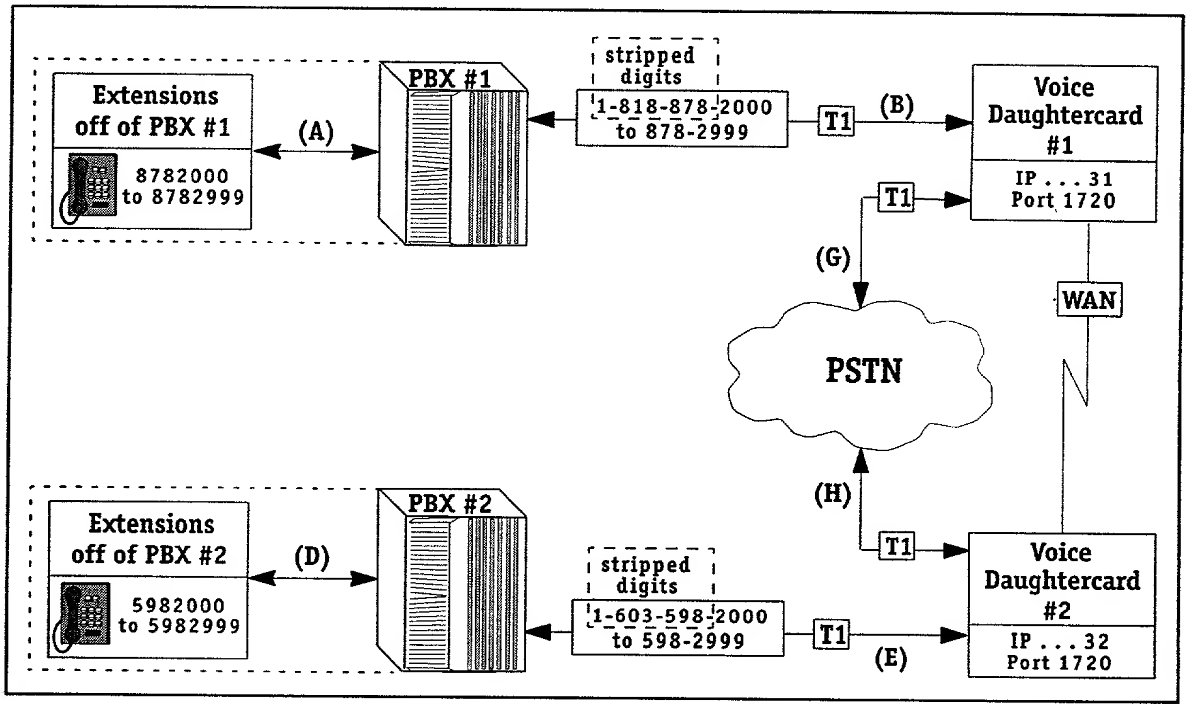
LEGEND for Diagram Components	
PBX #1 Configuration	**PBX #2 Configuration**
— Expects to receive four digits on trunk **(B)** and then uses these digits to route calls to lines **(A)** or trunk **(G)**.	— Expects to receive four digits on **(E)** and then uses these digits to route calls to lines **(D)** or trunk **(H)**.
— Routes VoIP calls to trunk **(F)** and sends four digits to voice daughtercard.	— Routes VoIP calls to trunk **(C)** and sends four digits to voice daughtercard.
— Routes all 1-1-npa-nxx-xxxx PSTN calls to trunk **(G)**.	— Routes all 1-1-npa-nxx-xxxx PSTN calls to trunk **(H)**.
— Routes all PSTN calls to trunk **(G)**.	— Routes all PSTN calls to trunk **(H)**.

VoIP Networks with PSTN — Example 15

Strip Digit Length (4)

This dialing scheme is used to strip digits from an NANP call.

Any number used as a site prefix cannot be used for the first digit of any valid extension.



Example 15 — Strip Digit Length (4)

LEGEND for Diagram Components	
PBX #1 Configuration	**PBX #2 Configuration**
— Expects to receive seven digits on trunk **(B)** and then uses these digits to route calls to lines **(A)** or trunk **(G)**.	— Expects to receive seven digits on trunk **(E)** and then uses these digits to route calls to lines **(D)** or trunk **(H)**.
— Routes VoIP calls to trunk **(F)** and sends four digits to voice daughtercard.	— Routes VoIP calls to trunk **(C)** and sends four digits to voice daughtercard.
— Routes all 1-1-npa-nxx-xxxx PSTN calls to trunk **(G)**.	— Routes all 1-1-npa-nxx-xxxx PSTN calls to trunk **(H)**.
— Routes all PSTN calls to trunk **(G)**.	— Routes all PSTN calls to trunk **(H)**.

Remarks

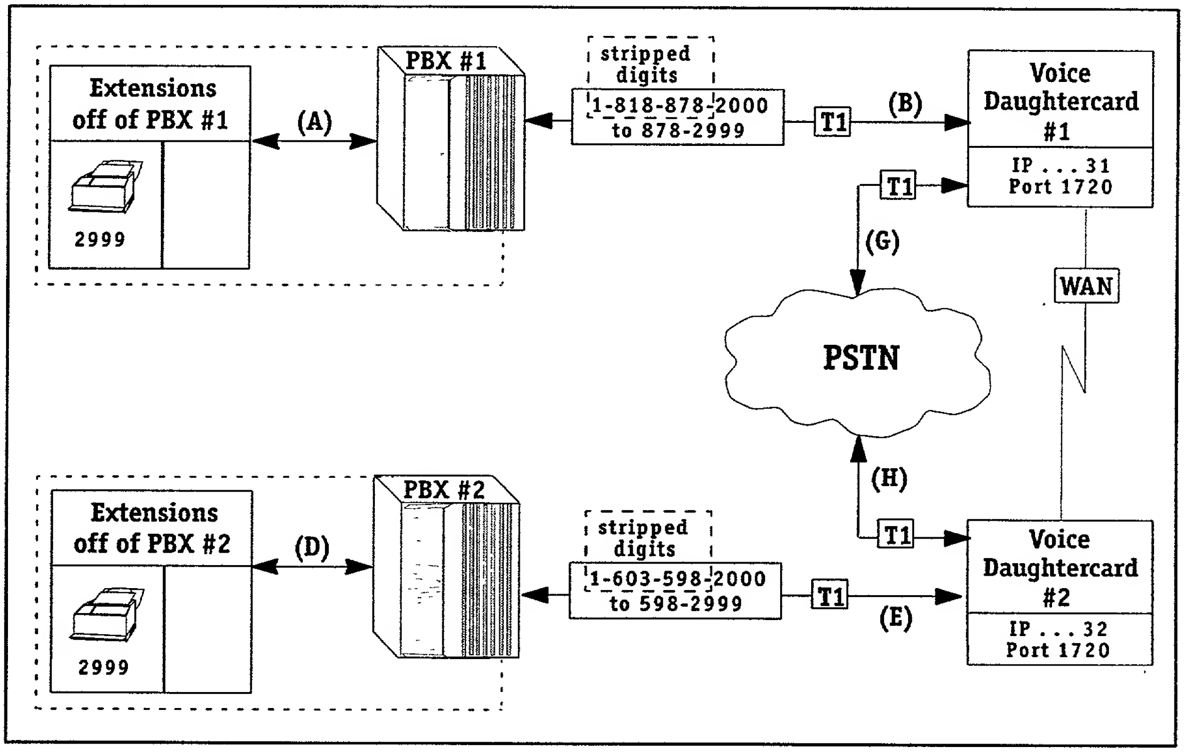
Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard **(A)**	*voice destination local channel* (page 5-244)
Local channel — two hunt groups (24 channels per group/T1) **(F)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Site prefix — no site prefix **(I)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Voice phone group type — eleven digit local extensions **(M)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — NANP extensions **(N)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — INTL extension **(O)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — PSTN NANP **(P)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — PSTN INTL **(Q)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Strip digit length — 4 **(U)**	*voice phone group strip digit length* (page 5-257)

VoIP Networks with PSTN — Example 16

FAX over IP Network

This dialing scheme is used to create a Fax Over IP network.



Example 16 — Fax over IP Network

LEGEND for Diagram Components	
PBX #1 Configuration	**PBX #2 Configuration**
— Expects to receive four digits on trunk **(B)** and then uses these digits to route calls to lines **(A)** or trunk **(G)**.	— Expects to receive four digits on trunk **(E)** and then uses these digits to route calls to lines **(D)** or trunk **(H)**.
— Routes VoIP calls to trunk **(F)** and sends four digits to voice daughtercard.	— Routes VoIP calls to trunk **(C)** and sends four digits to voice daughtercard.
— Routes all 1-1-npa-nxx-xxxx PSTN calls to trunk **(G)**.	— Routes all 1-1-npa-nxx-xxxx PSTN calls to trunk **(H)**.
— Routes all PSTN calls to trunk **(G)**.	— Routes all PSTN calls to trunk **(H)**.

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard **(A)**	*voice destination local channel* (page 5-244)
H.323 gateway to H.323 gatekeeper (RADVision) **(B)**	*voice network h.323 gatekeeper control* (page 5-230) *voice network h.323 gatekeeper mode* (page 5-231) *voice network h.323 gatekeeper address* (page 5-232)
H.323 gateway to Microsoft NetMeeting (without FastStart) **(C)**	*voice destination h.323 endpoint* (page 5-243)
Local channel — individual hunt groups (48 channels per group/T1) **(D)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Local channel — four hunt groups (12 channels per group/T1) **(E)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Local channel — two hunt groups (24 channels per group/T1) **(F)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Local channel — one hunt group (48 channels across two T1s) **(G)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Local channel — two hunt groups (60 channels across two E1s) **(H)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Site prefix — no site prefix **(I)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Site prefix — single digit **(J)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Site prefix — multiple digits **(J1)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Voice phone group type — three digit local extensions **(K)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — four digit local extensions **(L)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — eleven digit local extensions **(M)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — NANP extensions **(N)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)

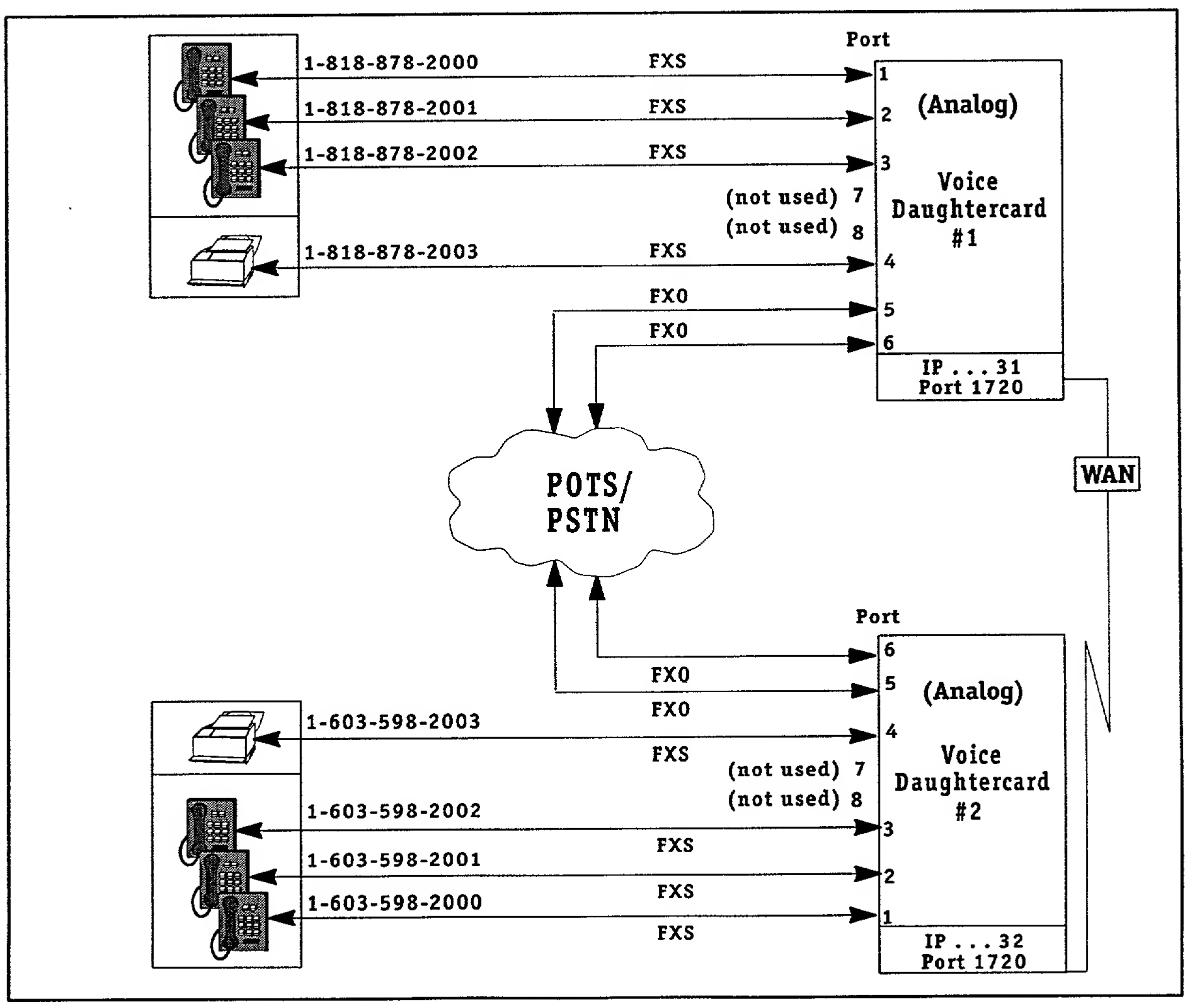
Features Supported	Primary CLI Commands Used
Voice phone group type — INTL extension **(O)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — PSTN NANP **(P)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — PSTN INTL **(Q)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Strip digit length — no strip digits **(R)**	*voice phone group strip digit length* (page 5-257)
Strip digit length — 1 **(S)**	*voice phone group strip digit length* (page 5-257)
Strip digit length — 2 **(T)**	*voice phone group strip digit length* (page 5-257)
Strip digit length — 4 **(U)**	*voice phone group strip digit length* (page 5-257)
Strip digit length — 7 **(V)**	*voice phone group strip digit length* (page 5-257)

VoIP Networks with PSTN — Example 17

Caller ID (Forwarding)

This dialing scheme is used to provide Caller ID forwarding of a caller's name and phone number to the called party. It requires analog connections using Plain Old Telephone Service (POTS) Foreign Exchange Office (FXO) and Foreign Exchange Station (FXS) signaling. It requires installation of analog voice switching grand-daughtercards (VSA FXS or FXO). See Chapter 2, "VoIP Daughtercards" for a description of the various daughtercards, and Chapter 4, "Setup and Installation" for details on installation. *Caller ID forwarding is not available this release.*

The VSA daughtercard does not provide call transfer, call hold or call forwarding. Each analog daughtercard has eight ports. In this example, a total of six lines are being used; four are FXS and two are FXO. Ports 7 and 8 are not used. FXS connections go to the physical phone, answering maching, fax or modem (1440 baud only); each device must have it's own unique telephone number. FXO lines go to the outside (POTS PSTN) via the voice daughtercard. No fractional hunt groups or trunks are used in this dialing scheme. The Caller ID information received on FXO lines is automatically forwarded to the FXS lines. Dialing schemes No. 17 and 18 (Caller ID, Forwarding and Static) apply only to VSD E1 or VSA daughtercards, or VSD T1 when configured for FXS Loop Start signaling.



Example 17 — Caller ID (Forwarding)

LEGEND for Diagram Components	
PBX #1 Configuration	**PBX #2 Configuration**
— Expects to receive four digits on trunk **(B)** and then uses these digits to route calls to lines **(A)** or trunk **(G)**.	— Expects to receive four digits on trunk **(E)** and then uses these digits to route calls to lines **(D)** or trunk **(H)**.
— Routes VoIP calls to trunk **(F)** and sends four digits to voice daughtercard.	— Routes VoIP calls to trunk **(C)** and sends four digits to voice daughtercard.
— Routes all 1-1-npa-nxx-xxxx PSTN calls to trunk **(G)**	— Routes all 1-1-npa-nxx-xxxx PSTN calls to trunk **(H)**.
— Routes all PSTN calls to trunk **(G)**.	— Routes all PSTN calls to trunk **(H)**.

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard **(A)**	*voice destination local channel* (page 5-244)
H.323 gateway to H.323 gatekeeper (RADVision) **(B)**	*voice network h.323 gatekeeper control* (page 5-230) *voice network h.323 gatekeeper mode* (page 5-231) *voice network h.323 gatekeeper address* (page 5-232)
H.323 gateway to Microsoft NetMeeting (without FastStart) **(C)**	*voice destination h.323 endpoint* (page 5-243)
Local channel — individual hunt groups (48 channels per group/T1) **(D)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Local channel — four hunt groups (12 channels per group/T1) **(E)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Local channel — two hunt groups (24 channels per group/T1) **(F)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Local channel — one hunt group (48 channels across two T1s) **(G)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Local channel — two hunt groups (60 channels across two E1s) **(H)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Site prefix — no site prefix **(I)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Site prefix — single digit **(J)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Site prefix — multiple digits **(J1)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)

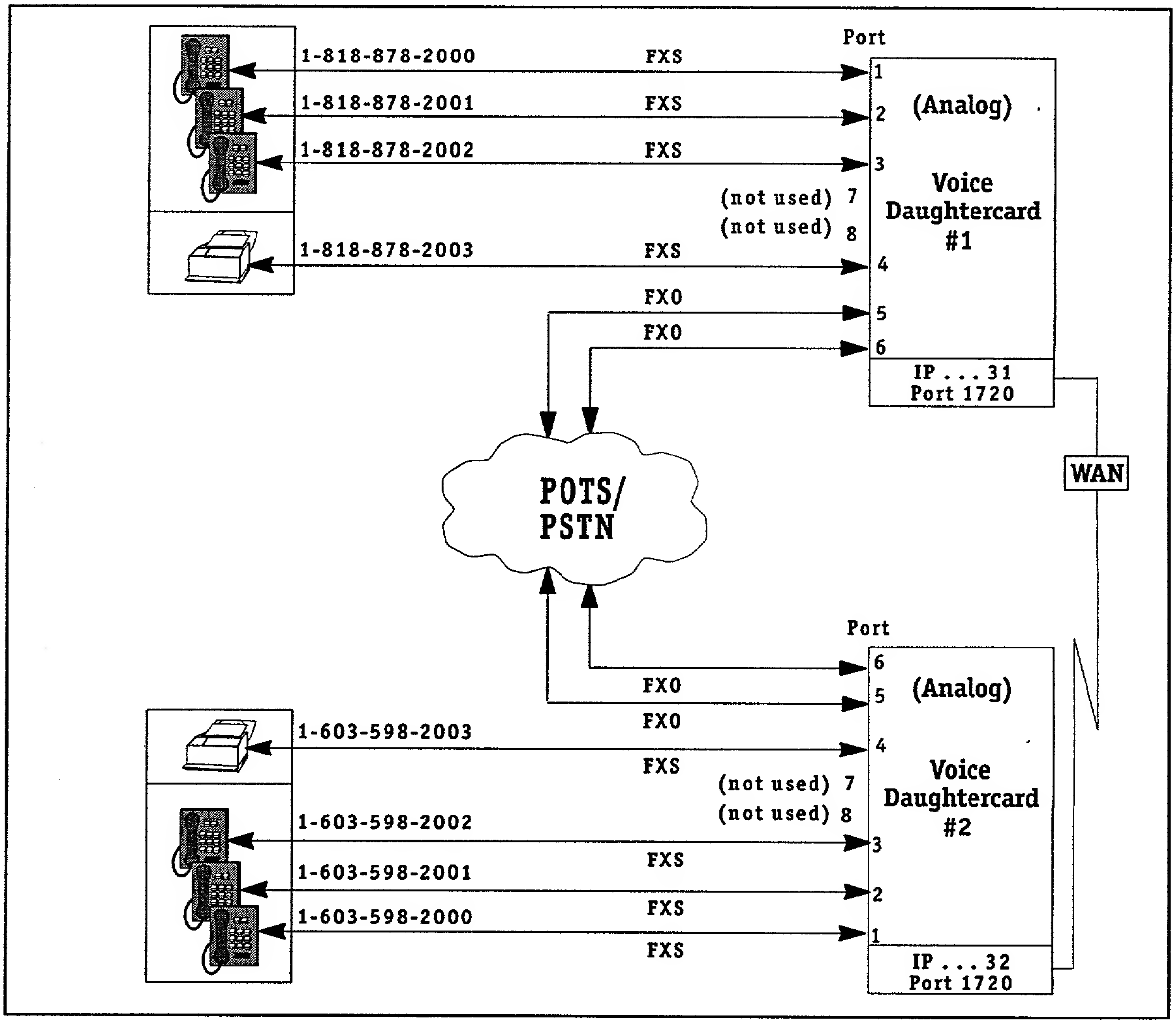
Features Supported	Primary CLI Commands Used
Voice phone group.type — three digit local extensions **(K)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — four digit local extensions **(L)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — eleven digit local extensions **(M)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — NANP extensions **(N)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — INTL extension **(O)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — PSTN NANP **(P)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — PSTN INTL **(Q)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Strip digit length — no strip digits **(R)**	*voice phone group strip digit length* (page 5-257)
Strip digit length — 1 **(S)**	*voice phone group strip digit length* (page 5-257)
Strip digit length — 2 **(T)**	*voice phone group strip digit length* (page 5-257)
Strip digit length — 4 **(U)**	*voice phone group strip digit length* (page 5-257)
Strip digit length — 7 **(V)**	*voice phone group strip digit length* (page 5-257)

VoIP Networks with PSTN — Example 18

Caller ID (Static)

This dialing scheme is very similar to Example 17, Caller ID (Forwarding) with the main difference being that Static Caller ID generates the same Caller ID name and number, and always overrides inbound FXO Caller ID name and number. See Chapter 2, "VoIP Daughtercards" for a description of the various daughtercards, and Chapter 4, "Setup and Installation" for details on installation.

Dialing schemes No. 17 and 18 (Caller ID, Forwarding and Static) apply only to VSD E1 or VSA daughtercards, or VSD T1 when configured for FXS Loop Start signaling.



Example 18 — Caller ID (Static)

LEGEND for Diagram Components	
PBX #1 Configuration	**PBX #2 Configuration**
— Expects to receive four digits on trunk **(B)** and then uses these digits to route calls to lines **(A)** or trunk **(G)**.	— Expects to receive four digits on trunk **(E)** and then uses these digits to route calls to lines **(D)** or trunk **(H)**.
— Routes VoIP calls to trunk **(F)** and sends four digits to voice daughtercard	— Routes VoIP calls to trunk **(C)** and sends four digits to voice daughtercard.
— Routes all 1-1-npa-nxx-xxxx PSTN calls to trunk **(G)**.	— Routes all 1-1-npa-nxx-xxxx PSTN calls to trunk **(H)**.
— Routes all PSTN calls to trunk **(G)**.	— Routes all PSTN calls to trunk **(H)**.

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

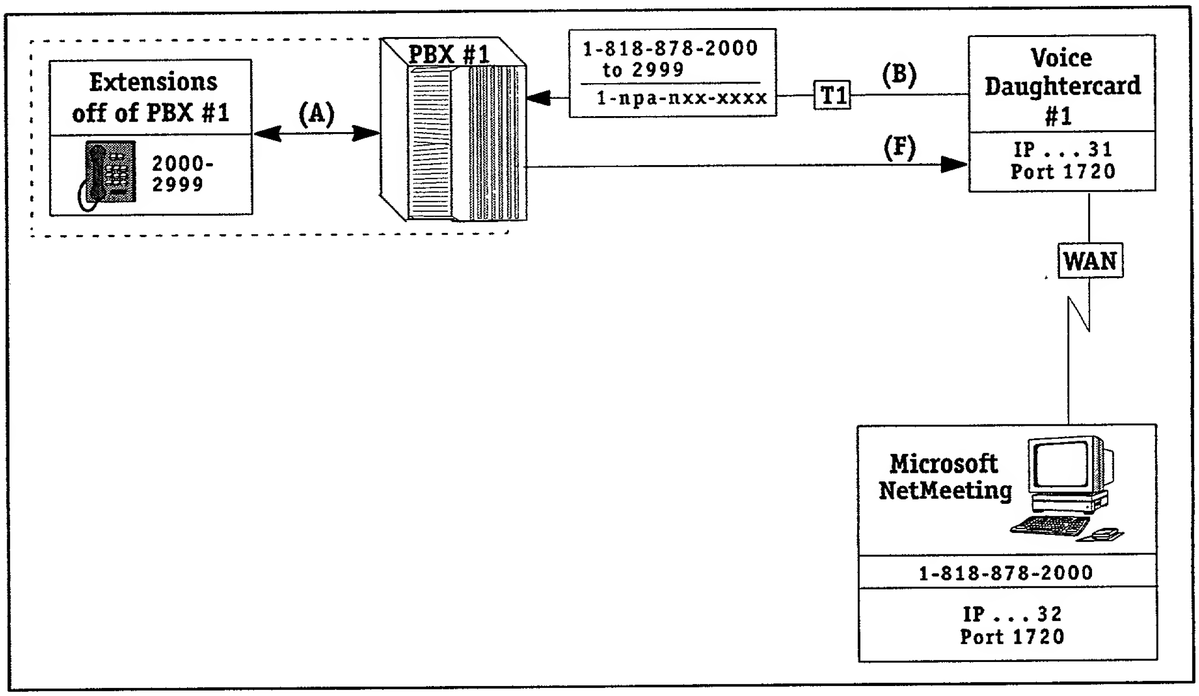
Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard **(A)**	*voice destination local channel* (page 5-244)
H.323 gateway to H.323 gatekeeper (RADVision) **(B)**	*voice network h.323 gatekeeper control* (page 5-230) *voice network h.323 gatekeeper mode* (page 5-231) *voice network h.323 gatekeeper address* (page 5-232)
H.323 gateway to Microsoft NetMeeting (without FastStart) **(C)**	*voice destination h.323 endpoint* (page 5-243)
Local channel — individual hunt groups (48 channels per group/T1) **(D)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Local channel — four hunt groups (12 channels per group/T1) **(E)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Local channel — two hunt groups (24 channels per group/T1) **(F)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Local channel — one hunt group (48 channels across two T1s) **(G)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Local channel — two hunt groups (60 channels across two E1s) **(H)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Site prefix — no site prefix **(I)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Site prefix — single digit **(J)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Site prefix — multiple digits **(J1)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Voice phone group type — three digit local extensions **(K)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — four digit local extensions **(L)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — eleven digit local extensions **(M)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — NANP extensions **(N)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)

Features Supported	Primary CLI Commands Used
Voice phone group type — INTL extension **(O)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — PSTN NANP **(P)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — PSTN INTL **(Q)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Strip digit length — no strip digits **(R)**	*voice phone group strip digit length* (page 5-257)
Strip digit length — 1 **(S)**	*voice phone group strip digit length* (page 5-257)
Strip digit length — 2 **(T)**	*voice phone group strip digit length* (page 5-257)
Strip digit length — 4 **(U)**	*voice phone group strip digit length* (page 5-257)
Strip digit length — 7 **(V)**	*voice phone group strip digit length* (page 5-257)

VoIP Networks with Interoperability — Example 19

H.323 Gateway to Microsoft NetMeeting (without FastStart)

This dialing scheme example is used to connect a voice daughtercard to a Microsoft NetMeeting H.323-compliant terminal with microphone.



Example 19 — H.323 Gateway to Microsoft NetMeeting

LEGEND for Diagram Components	
PBX #1 Configuration	**PBX #2 Configuration**
— Expects to receive four digits on trunk **(B)** and then uses these digits to route calls to lines **(A)** or trunk **(G)**.	— Expects to receive four digits on trunk **(E)** and then uses these digits to route calls to lines **(D)** or trunk **(G)**.
— Routes VoIP calls to trunk **(F)** and sends four digits to voice daughtercard.	— Routes VoIP calls to trunk **(C)** and sends four digits to voice daughtercard.
— Routes all 1-1-npa-nxx-xxxx PSTN calls to trunk **(G)**.	— Routes all 1-1-npa-nxx-xxxx PSTN calls to trunk **(H)**.
— Routes all PSTN calls to trunk **(G)**.	— Routes all PSTN calls to trunk **(H)**.

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

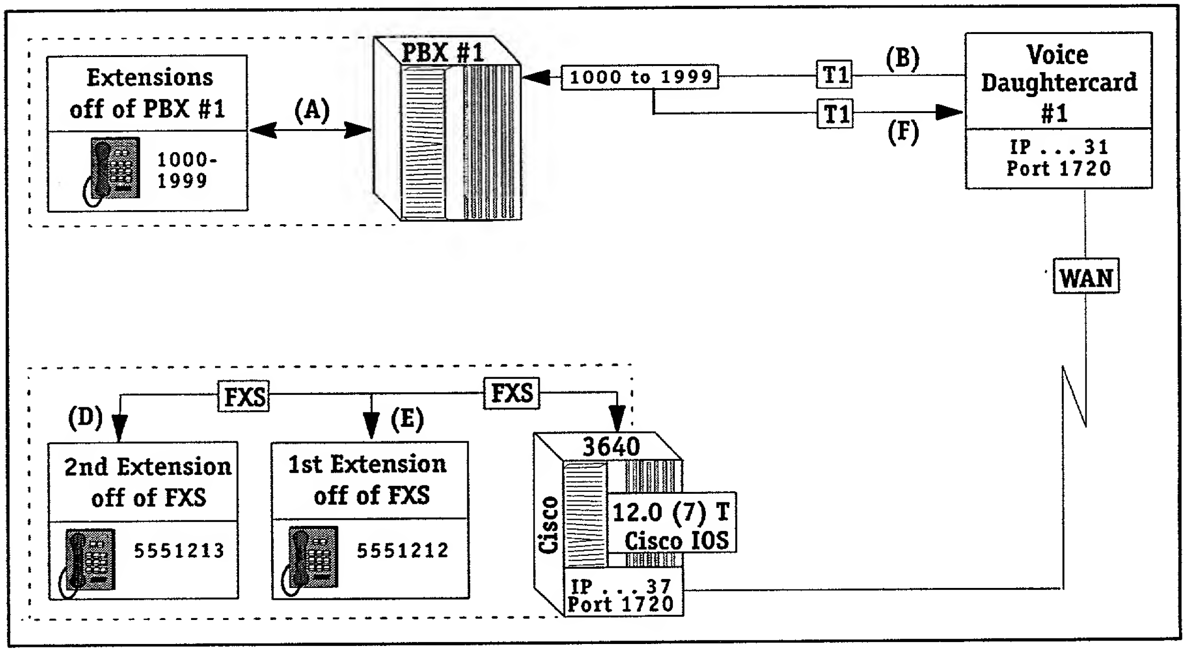
To connect to Microsoft NetMeeting the voice coding profile must be set to (tbd), the codec type set to g.(tbd), the VIF set to 10, and the VPI set to (tbd) ms.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard **(A)**	*voice destination local channel* (page 5-244)
Local channel — two hunt groups (24 channels per group/T1) **(F)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Site prefix — no site prefix **(I)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Voice phone group type — NANP extensions **(N)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — INTL extension **(O)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — PSTN NANP **(P)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — PSTN INTL **(Q)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Strip digit length — no strip digits **(R)**	*voice phone group strip digit length* (page 5-257)
Strip digit length — 1 **(S)**	*voice phone group strip digit length* (page 5-257)
Strip digit length — 2 **(T)**	*voice phone group strip digit length* (page 5-257)

VoIP Networks with Interoperability — Example 20

H.323 Gateway to Cisco Router

This dialing scheme is used to connect a voice daughtercard to a Cisco 3540 router running Cisco IOS (7) 12.0.



Example 20 — H.323 Gateway to Cisco Router

LEGEND for Diagram Components	
PBX #1 Configuration	**PBX #2 Configuration**
— Expects to receive four digits on trunk **(B)**, and then uses these digits to route calls to lines **(A)** or trunk **(G)**.	— Expects to receive four digits on trunk **(E)**, and then uses these digits to route calls to lines **(D)** or trunk **(H)**.
— Routes VoIP calls to trunk **(F)** and sends four digits to voice daughtercard.	— Routes VoIP calls to trunk **(C)** and sends four digits to voice daughtercard.
— Routes all 1-1-npa-1-1-npa-nxx-xxxx PSTN calls to trunk **(G)**.	— Routes all 1-1-npa-nxx-xxxx PSTN calls to trunk **(H)**.
— Routes all PSTN calls to trunk **(G)**.	— Routes all PSTN calls to trunk **(H)**.

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

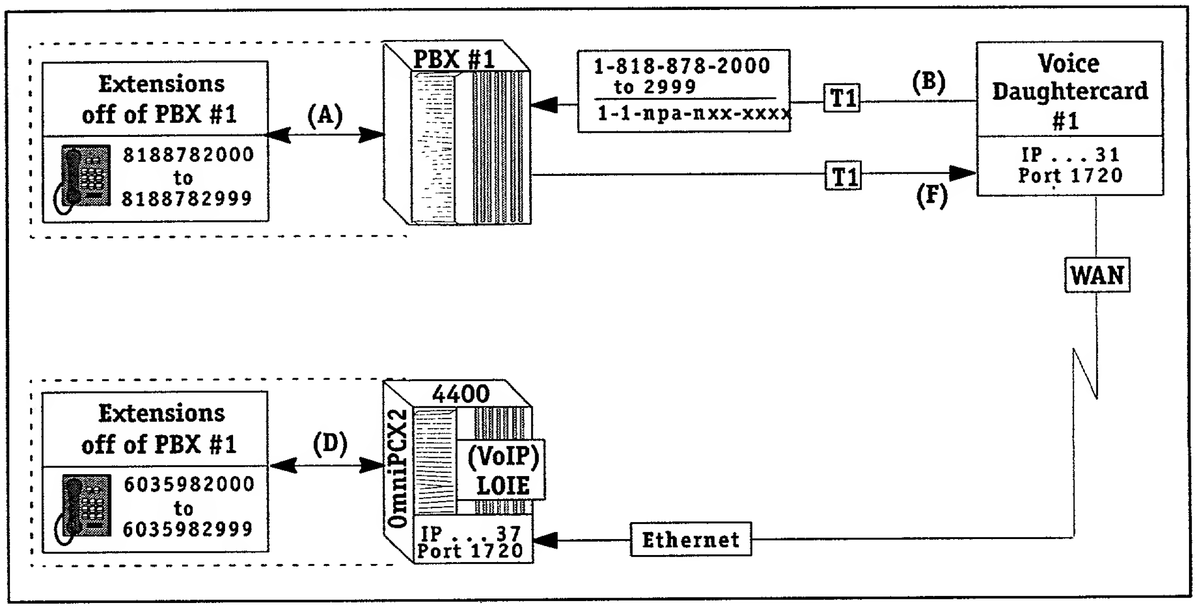
To connect to a Cisco Router to a voice daughtercard, the voice coding profile must be set to (tbd), the codec type set to g.(tbd), the VIF set to 10, and the VPI (voice packet interval) set to (tbd) ms.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard **(A)**	*voice destination local channel* (page 5-244)
Local channel — two hunt groups (24 channels per group/T1) **(F)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Site prefix — no site prefix **(I)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Voice phone group type — three digit local extensions **(K)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — four digit local extensions **(L)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — eleven digit local extensions **(M)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Strip digit length — 4 **(U)**	*voice phone group strip digit length* (page 5-257)
Strip digit length — 7 **(V)**	*voice phone group strip digit length* (page 5-257)

VoIP Networks with Interoperability — Example 21

H.323 Gateway to OmniPCX 4400

This dialing scheme is used to connect an H.323 Gateway to an OmniPCX 4400, and requires full compliance with H.323 Version 1. Only H.323 V1 voice capability is supported by the OmniPCX. The OmniPCX does not support Codec Fax or Codec FAX T.38. The OmniPCX 4400 also requires installation of an LIOE voice card to provide VoIP. Refer to the Alcatel OmniPCX 4400 Operations Manual for more information.



Example 21 — H.323 Gateway to OmniPCX 4400

LEGEND for Diagram Components	
PBX #1 Configuration	**OmniPCX #2 Configuration**
— Expects to receive four digits on trunk **(B)** and then uses these digits to route calls to lines **(A)** or trunk **(G)**.	— Expects to receive h.323 V. 1 calls on LIOE ethernet port, and then uses embedded h.323 information to route calls to lines **(D)**.
— Routes VoIP calls to trunk **(F)** and sends four digits to voice daughtercard.	— Routes VoIP calls to trunk **(C)** and sends four digits to voice daughtercard.
— Routes all 1-npa-nxx-xxxx PSTN calls to trunk **(G)**.	— Routes all 1-npa-nxx-xxxx PSTN calls to trunk **(H)**.
— Routes all PSTN calls to trunk **(G)**.	— Routes all PSTN calls to trunk **(H)**.

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

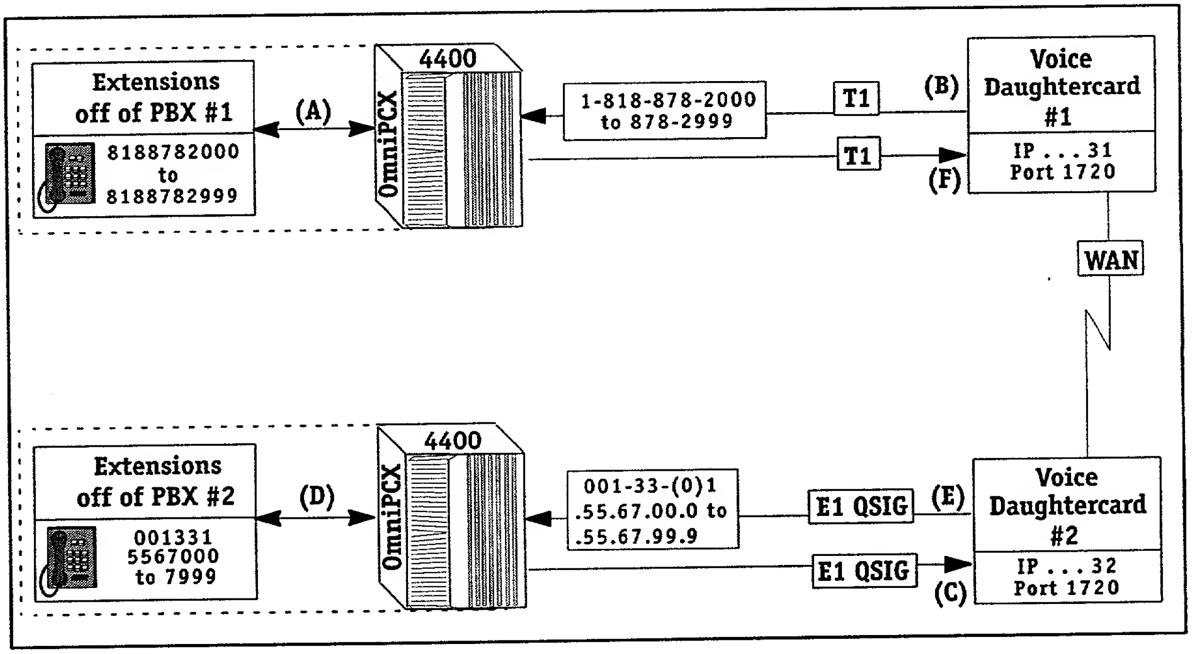
To connect to the LIOE card on the OmniPCX 4400, the voice coding profile must be set to (tbd), the codec type set to g.(tbd), the VIF set to 10, and the VPI (voice packet interval) set to (tbd) ms.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard **(A)**	*voice destination local channel* (page 5-244)
Local channel — individual hunt groups (48 channels per group/T1) **(D)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Site prefix — no site prefix **(I)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Site prefix — single digit **(J)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Voice phone group type — four digit local extensions **(L)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — eleven digit local extensions **(M)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Strip digit length — no strip digits **(R)**	*voice phone group strip digit length* (page 5-257)

VoIP Networks with Interoperability — Example 22

OmniPCX 4400 and E1 QSIG

This dialing scheme is used to connect an OmniPCX 4400 to an E1 QSIG port on a voice daughtercard. *Not available this release.*



Example 22 — OmniPCX 4400 and E1 QSIG

LEGEND for Diagram Components	
OmniPCX #1 Configuration	**OmniPCX#2 Configuration**
— Expects to receive four digits on trunk **(B)** and then uses these digits to route calls to lines **(A)** or trunk **(G)**.	— Expects to receive four digits on trunk **(E)** and then uses these digits to route calls to lines **(D)** or trunk **(H)**.
— Routes VoIP calls to trunk **(F)** and sends four digits to voice daughtercard.	— Routes VoIP calls to trunk **(C)** and sends four digits to voice daughtercard.
— Routes all 1-npa-nxx-xxxx PSTN calls to trunk **(G)**.	— Routes all 1-npa-nxx-xxxx PSTN calls to trunk **(H)**.
— Routes all PSTN calls to trunk **(G)**.	— Routes all PSTN calls to trunk **(H)**.

Remarks

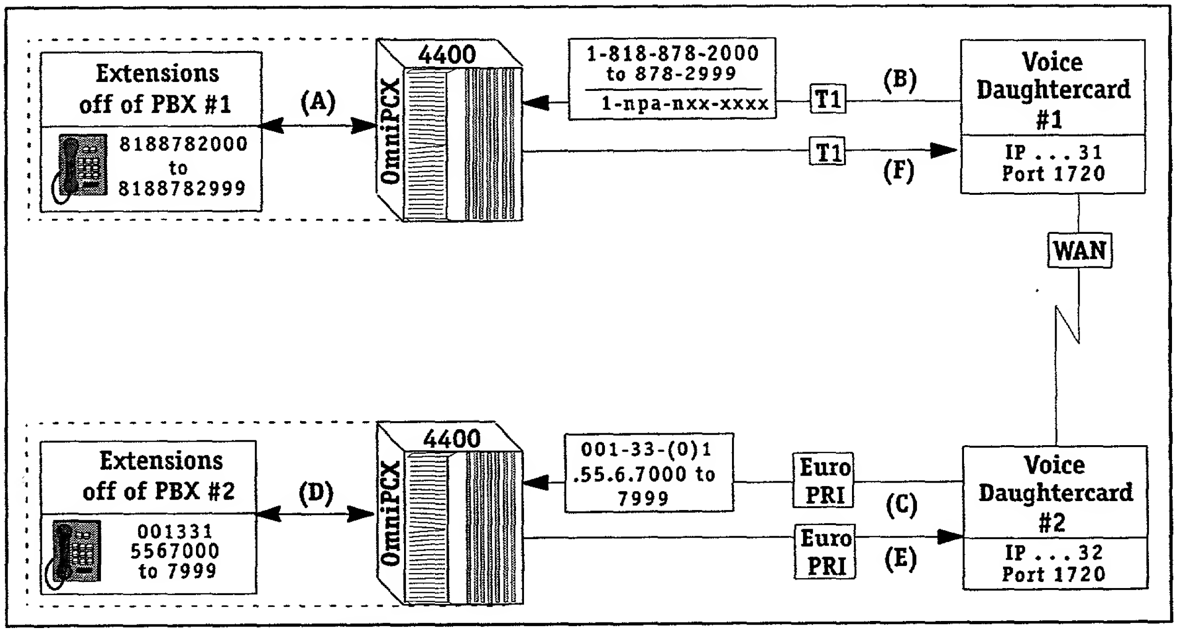
Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard **(A)**	*voice destination local channel* (page 5-244)
Local channel — two hunt groups (24 channels per group/T1) **(F)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Local channel — two hunt groups (60 channels across two E1s) **(H)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Site prefix — no site prefix **(I)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Voice phone group type — eleven digit local extensions **(M)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Strip digit length — no strip digits **(R)**	*voice phone group strip digit length* (page 5-257)
Digital Interface type — T1 **(W)**	*voice port interface type* (page 5-31)
Digital Interface type — E1 (QSIG) **(X)**	*voice port interface type* (page 5-31)

VoIP Networks with Interoperability — Example 23

OmniPCX and Euro PRI

This dialing scheme is used to connect a voice daughtercard to an OmniPCX using Euro PRI.



Example 23 — OmniPCX 4400 and Euro PRI

LEGEND for Diagram Components	
PBX #1 Configuration	**PBX #2 Configuration**
— Expects to receive four digits on trunk **(B)** and then uses these digits to route calls to lines **(A)** or trunk **(G)**.	— Expects to receive four digits on trunk **(E)** and then uses these digits to route calls to lines **(D)** or trunk **(H)**.
— Routes VoIP calls to trunk **(F)** and sends four digits to voice daughtercard.	— Routes VoIP calls to trunk **(C)** and sends four digits to voice daughtercard.
— Routes all 1-npa-nxx-xxxx PSTN calls to trunk **(G)**.	— Routes all 1-npa-nxx-xxxx PSTN calls to trunk **(H)**.
— Routes all PSTN calls to trunk **(G)**.	— Routes all PSTN calls to trunk **(H)**.

Remarks

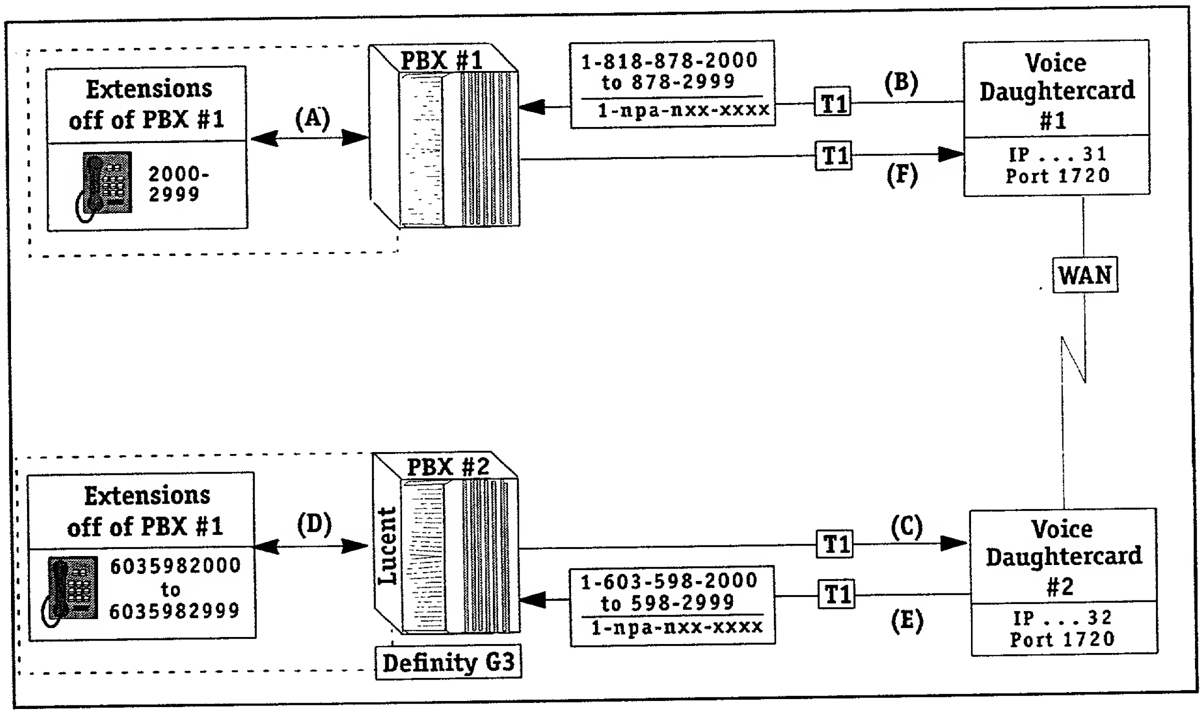
Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard **(A)**	*voice destination local channel* (page 5-244)
Local channel — two hunt groups (24 channels per group/T1) **(F)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Site prefix — no site prefix **(I)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Voice phone group type — eleven digit local extensions **(M)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Strip digit length — no strip digits **(R)**	*voice phone group strip digit length* (page 5-257)

VoIP Networks with Interoperability — Example 24

Other PBXs

This dialing scheme is used to connect a voice daughtercard to a Lucent Definity G3 running the following operating system: System G3siV4, Software Version G34.i.04.0.054.0.



Example 24 — Other PBXs

LEGEND for Diagram Components	
PBX #1 Configuration	**PBX #2 Configuration**
— Expects to receive four digits on trunk **(B)** and then uses these digits to route calls to lines **(A)** or trunk **(G)**.	— Expects to receive four digits on trunk **(E)** and then uses these digits to route calls to lines **(D)** or trunk **(H)**.
— Routes VoIP calls to trunk **(F)** and sends four digits to voice daughtercard.	— Routes VoIP calls to trunk **(C)** and sends four digits to voice daughtercard.
— Routes all 1-npa-nxx-xxxx PSTN calls to trunk **(G)**.	— Routes all 1-npa-nxx-xxxx PSTN calls to trunk **(H)**.
— Routes all PSTN calls to trunk **(G)**.	— Routes all PSTN calls to trunk **(H)**.

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard **(A)**	*voice destination local channel* (page 5-244)
Local channel — two hunt groups (24 channels per group/T1) **(F)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Site prefix — no site prefix **(I)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Voice phone group type — eleven digit local extensions **(M)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Strip digit length — no strip digits **(R)**	*voice phone group strip digit length* (page 5-257)

To follow

Master List of Features by CLI Command

Refer to the table below for a list of VoIP features, along with the most common commands used for setting up each feature. Corresponding page numbers for the *VoIP CLI Commands* chapter are also included.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard **(A)**	*voice destination local channel* (page 5-244)
H.323 gateway to H.323 gatekeeper (RADVision) **(B)**	*voice network h.323 gatekeeper control* (page 5-230) *voice network h.323 gatekeeper mode* (page 5-231) *voice network h.323 gatekeeper address* (page 5-232)
H.323 gateway to Microsoft NetMeeting (without FastStart) **(C)**	*voice destination h.323 endpoint* (page 5-243)
Local channel — individual hunt groups (48 channels per group/T1) **(D)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Local channel — four hunt groups (12 channels per group/T1) **(E)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Local channel — two hunt groups (24 channels per group/T1) **(F)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Local channel — one hunt group (48 channels across two T1s) **(G)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Local channel — two hunt groups (60 channels across two E1s) **(H)**	*voice numbering plan hunt method* (page 5-268) *voice numbering plan destination member* (page 5-270) *voice numbering plan phone group member* (page 5-271)
Site prefix — no site prefix **(I)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Site prefix — single digit **(J)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Site prefix — multiple digits **(J1)**	*voice phone group site prefix* (page 5-251) *voice phone group site prefix digits* (page 5-252)
Voice phone group type — three digit local extensions **(K)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — four digit local extensions **(L)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)

Features Supported	Primary CLI Commands Used
Voice phone group type — eleven digit local extensions **(M)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — NANP extensions **(N)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — INTL extension **(O)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — PSTN NANP **(P)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Voice phone group type — PSTN INTL **(Q)**	*voice phone group type* (page 5-253) *voice phone group format* (page 5-256) *voice phone group add numbers* (page 5-261)
Strip digit length — no strip digits **(R)**	*voice phone group strip digit length* (page 5-257)
Strip digit length — 1 **(S)**	*voice phone group strip digit length* (page 5-257)
Strip digit length — 2 **(T)**	*voice phone group strip digit length* (page 5-257)
Strip digit length — 4 **(U)**	*voice phone group strip digit length* (page 5-257)
Strip digit length — 7 **(V)**	*voice phone group strip digit length* (page 5-257)
Digital Interface type — T1 **(W)**	*voice port interface type* (page 5-31)
Digital Interface type — E1 (QSIG) **(X)**	*voice port interface type* (page 5-31)
Digital Interface type — E1 ISDN PRI (Euro PRI) **(Y)**	*voice port interface type* (page 5-31)
Digital Interface type — BRI Euro **(Z)**	*voice port interface type* (page 5-31)
Digital Interface type — FXS **(AA)**	*voice port interface type* (page 5-31)
Digital Interface type — FXO **(AB)**	*voice port interface type* (page 5-31)

5 VoIP Commands

The following chapter contains information on using the VoIP command line interface (CLI) to configure VoIP switches. The commands are divided into seven major categories consisting of the following subcategories as listed below.

Command Category	Command Subcategory
Voice Switching Daughtercards	• Dialing Timers • Gateway Mode • Voice Ports
Channel Properties	• Voice Channel Mode • PLAR (Private Line Automatic Ringdown) • Voice Channel Initialization
Telephony Signaling Templates	• General Signaling • Ear & Mouth (E&M) • Foreign Exchange Station (FXS) • Foreign Exchange Office (FXO) • Call Signaling Capabilities • Inbound/Outbound Caller ID • Call Progress Tones • Echo and Acoustic Echo Cancellers • Overrides for Call Signaling
Coding Profiles	• Codecs • Voice Mode Parameters • Voice Network Buffers • Voice Activity Detection • Tone Detection • Echo Canceller • Facsimile Modem • Facsimile T.38 Modem • Silence Detection • General Caller ID
Voice Network Templates	• H.323 Gateway Discovery • H.323 Gateway Configuration • H.323 Gateway Operations
Network Dialing Scheme	• Destinations • Phone Groups/Parameters • Numbering Plans/Hunt Method
System-Wide VoIP	• Voice Switching Daughtercard Parameters • General Telephony and Telephony Channels • Voice Play Out • Voice Daughtercard (DSP Receive and Transmit) • Errors • Modem, Fax and ISDN statistics

To use this chapter, refer to the command task list below to find the page number for a specific task. The commands use a simple, line-at-a-time prompt and response scheme. The CLI interface presents a single prompt character (depending on the operating system) at the beginning of each command line; however, this does not apply to the **vsmboot.asc file** in which there is no response. For details on the **vsmboot.asc** file and other similar type files, see Chapter 5, "Setup and Installation," and Appendix A, "VSM Boot Files," which contains example VoIP boot files.

♦ Note ♦

Do *not* use any CLI **view** commands from within the aforementioned VSM boot files at any time.

The CLI text-based commands used in VoIP are intended for use by Network Administrators and technical staff to configure Alcatel switches for VoIP. Commands are not case sensitive unless otherwise stated; however, if a name or string is used it will be case sensitive. Commands which may apply to either E1, E1 ISDN PRI, or BRI Euro, are collectively referenced in command names under E1.

Typically, command tasks which begin with "specify" have more than two parameters from which to choose, whereas command tasks beginning with "set" are generally an either/or type command, e.g. **on** or **off**.

When entering certain values such as slot, port and channel numbers in the command syntax, refer to the configuration table on the following page for valid entries.

Parameters

VoIP configuration parameters include system-wide configuration as well as per channel, per port, and per daughtercard configuration. System-wide configuration includes signaling templates, coding profiles, voice network templates (including H.323 gateway and gatekeeper configuration). Per channel configuration includes general channel and channel-level telephony signaling configuration. Many voice switching daughtercard parameters can also be applied using profiles and templates as described below.

Profiles and Templates

Profiles are parameters that define the way a device such as a VoIP H.323 gateway card acts. Once a profile, such as a coding profile has been assigned to a channel, it remains in effect until another profile is assigned. If any individual parameter of the profile is modified, it will take effect on all entities to which it had been defined the next the profile is requested from the voice switching daughtercard. *Coding profiles are not assigned to a physical entity because the coding profile used is determined at runtime.* If coding profile parameters are changed, the next time a modified coding profile is called by the switch, the new information will be obtained. Profiles are also used to associate specific entities (daughtercard with channels) on a voice switching daughtercard.

Templates are scripts that are assigned a set of parameters to define the way a physical entity, such as a chassis, slot, voice daughtercard, port or channel acts. After a template is assigned to an entity such as a voice daughtercard, it does not stay associated with that entity. Also, if any individual parameter of a template is modified, and then activated, it will not take effect until it is assigned to an entity, and previous assignment of the template to an entity is not changed.

Signaling templates, voice network templates, voice phone groups, voice destinations and voice numbering plans are all considered templates. On a channel level, templates can be assigned, added, and deleted. The last template assigned overrides any existing templates.

For a more detailed discussion of profiles and templates, see Chapter 2, "VoIP Features."

Parameter	Value	Range
Slot	**Physical**	**•Slot Number** Omni Access 512 value is always 4. Omni Switch/Router range is 2 to 3, 2 to 5, or 2 to 9 depending on chassis size.
Card	**Physical**	**•Card Number** OA 512 value is always 1. O S/R values are always 1 to 2.
Port	**Physical**	**•Port Number** VSD (digital) range is 1 to 2, or 1 to 4. VSB (digital; Euro ISDN BRI) range is 1 to 2 (OA 512), or 1 to 4 (OSR). VSA (analog) range is 1 to 2 (FXO), 1 to 4 (FXO/FXO or FXS), 1 to 6 (FXS and FXO), or 1 to 8 (FXS/FXS modules); applies only to OA 512. Same for O S/R except range is 1 to 16 based upon module installed.
Channel	**Logical**	**•Channel Number** VSD T1 range is 1 to 24. VSD E1 range is 1 to 30. VSB (Euro ISDN BRI) is 1 to 2. VSA (analog) value is always 1.
StartChannel-EndChannel	**Logical**	**•Range of Channels** Channel-to-Channel, e.g., 11-16. A channel number is assigned to every channel in the specified range; in this case, Channels 11, 12, 13, 14, 15 and 16.

Command Tasks

Voice Switching Daughtercard Commands

The commands listed and described below are used to configure and activate voice switching daughtercards, and include the following daughtercard functions: dialing timers, gateway mode, and voice ports.

Dialing timers determine how the DSPs on the daughtercard detect digits, and the ports provide an interface from the switch to the PSTN via a PBX or key system. The H.323 gateway enables or disables communications between the switch, the PSTN, and the PBX or key system.

Voice Switching Daughtercard (Activate)

assign IP address mask to voice switching daughtercard

assign IP address to voice switching daughtercard

assign default gateway to voice switching daughtercard

activate voice switching daughtercard

activate voice switching daughtercard *configuration*

save current text-based configuration to MPM flash (global "Save All" dump command)

H.323 Gateway Configuration and Runtime Parameters

outgoing faststart mode for gateway (on/off)

incoming faststart mode for gateway (on/off)

automatic answer for gateway (on/off)

Dialing Timers

first digit wait duration

inter digit wait duration

dial time wait duration

termination digit (optional)

Digital Port Configuration

Port Connection Type

voice port digital connection interface type (T1/E1/E1 ISDN PRI/BRI Euro)

Telephony Interface

voice port frame format

voice port circuit identifier (optional)

E1 voice port NFAS (enable/disable)

Line Build Out

voice port line haul (short haul/long haul)

T1 voice port line length

T1 voice port attenuation

E1 voice port cable type

Line Coding

voice port line coding

Facilities Data Link

T1 voice port facilities data link protocol

T1 voice port facilities data link port role (network/user)

Transmit Clock Source

voice port transmit clock source

Loop Back Mode

T1 voice port *loop back configuration*

Signaling Mode

voice port channel signaling mode

Trap Generation

E1 voice port trap generation (enable/disable)

Send Code

T1 voice port *loop back configuration to send (Not available this release.)*

ISDN

E1 ISDN port connection protocol (net/user/qmaster/qslave)

E1 ISDN connection switch type (net3/net5)

ISDN Control and Bearer Channels

E1 ISDN control (Data or "D") channels

E1 ISDN bearer ("B") channels

voice daughter card ip mask

Command Usage

Assign IP address mask to voice switching daughtercard.

Syntax Options

voice daughter card <*slot/card_number*> **ip mask** <*ip_address*>

Definitions:

slot	Specifies the chassis slot number where VSM is installed, (e.g., **2**).
card_number	Specifies the voice daughtercard position number, (e.g., **1**).
ip_address	Identifies the voice switching daughtercard by IP address mask, (e.g., **255.255.255.0**).

Default:
None

Command Examples:
voice daughter card 2/1 ip address mask 255.255.255.0
voice daughter card 2/2 ip address mask 255.255.255.0

Remarks

This command is required and must be included in the master vsmboot.asc file as per Chapter 5, "Setup and Installation."

voice daughter card ip address

Command Usage

Assign IP address to voice switching daughtercard.

Syntax Options

voice daughter card <*slot/card_number*> ip address <*ip_address*>

Definitions:

slot	Specifies the chassis slot number where VSM is installed, (e.g., **2**).
card_number	Specifies the voice daughtercard position number, (e.g., **1**).
ip_address	Identifies the voice switching daughtercard by IP address, (e.g., **127.0.0.0**).

Default:
None

Command Examples:
voice daughter card 2/1 ip address 127.0.0.0
voice daughter card 2/2 ip address 127.0.0.0

Remarks

This command is required and must be included in the master vsmboot.asc file as per Chapter 5, "Setup and Installation."

voice daughter card ip default gateway

Command Usage

Assign IP address default gateway to voice switching daughtercard.

Syntax Options

voice daughter card <*slot/card_number*> **ip default gateway** <*ip_address*>

Definitions:

slot	Specifies the chassis slot number where VSM is installed, (e.g., **2**).
card_number	Specifies the voice daughtercard position number, (e.g., **1**).
ip_address	Identifies the voice switching daughtercard by IP address default gateway, (e.g., **127.0.0.1**).

Default:
None

Command Examples:
voice daughter card 2/1 ip default gateway 127.0.0.1
voice daughter card 2/2 ip default gateway 127.0.0.1

voice daughter card activate

Command Usage

Activate previously configured voice switching daughtercard parameters.

Syntax Options

voice daughter card <*slot/port*> activate

Definitions:

slot	Specifies slot number of switching module installed in chassis, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).

♦ **Syntax Note** ♦

This command is always required, and must only be issued after all other relevant VoIP CLI commands have been configured. See Chapter 5, "Setup and Installation," for details regarding its use in the **vsmboot.asc** file.

This command does not apply to voice numbering plans, voice phone groups or voice destinations.

Before using this command, all channels should be placed out of service via the **voice channel state** command.

It is recommended that this command be used only during "off hours," or after connected devices are first instructed to stop routing calls to the card.

Default:
None

Command Examples:
voice daughter card 2/1 activate
voice daughter card 2/2 activate

Remarks

This command is required and must be included in the master vsmboot.asc file as per Chapter 5, "Setup and Installation."

This command activates all previously configured voice switching daughtercard parameters by transferring cached parameters to the daughtercard, and then placing the parameters into service from the card.

Using this command immediately puts all channels on the voice switching daughtercard out of service, shuts down the card, and automatically erases the previous configuration; this includes disconnecting any calls in progress, and ignoring any incoming traffic from either the phone or data networks.

Once the activate command is issued, the configuration should be saved to the flash directory on the switch via the global **dump** command. The **dump** command should only be used *after* a configured voice switching daughtercard has been activated, or an invalid configuration may be saved instead which may prevent the switch from booting properly the next time.

The **dump** command automatically creates the text file in the flash directory of the switch. To view the contents of the generated text file, you can use the **view file** command. For more information, refer to the **dump** and **view file** command descriptions in the *Command Line Reference Guide*.

dump

Command Usage

Save current text-based configuration to MPM flash (global "dump" save all command). The aggregate configuration is first captured and then saved to a single text file that can be viewed, edited, or reapplied to additional switches for implementation.

Syntax Options

dump {all | *feature-type* **} [file** *name* **]**

Definitions:

all	Specifies that information for all supported switch features (including VoIP) will be saved to the dump file.
feature-type	Specifies that only a particular feature will be saved to the dump file, (e.g., **voice**, **vlan**).
name	Specifies a user-defined name for the resulting dump file (18 characters maximum); (e.g., **snapshot1**, **vsmboot.asc**).

Default:
None

Command Examples:
dump all
dump voice
dump all file snapshot.1
dump voice file vsmboot.asc

Remarks

The **dump** command automatically creates the text file in the flash directory of the switch. To view the contents of the generated text file, you can use the **view file** command. For more information, refer to the **view file** command description in the *Command Line Reference Guide.*

Aside from basic system parameters, (e.g., system name), the **dump** command captures only *non-default* switch parameters for the specified switch features. For example, the default settings for VoIP will not be included in a dump file.

The text file can be edited using a standard text editor.

Screen Output

To view all voice daughtercard parameters, type **dump voice file** followed by valid vsm boot file name, e.g., **dump voice file vsmboot.asc.**, and then press **<Enter>**.

A screen similar to the following displays.

```
******************************************
***              Snap All            ***
******************************************
!
voice echo on
!
voice daughtercard 4/1 ip mask 255.255.255.0
!
voice daughtercard 4/1 ip address 127.0.0.0
!
!voice port 4/1 interface type T1
!
voice coding profile cp1
!
voice coding profile cp1 type pcm mulaw
!
voice channel 4/1/1 mode telephony
!
voice channel 4/1/2 mode telephony
!
voice channel 4/1/3 mode telephony
!
voice channel 4/1/4 mode telephony
!
voice channel 4/1/5 mode telephony
!
voice channel 4/1/6 mode telephony
!
voice channel 4/1/7 mode telephony
!
voice channel 4/1/8 mode telephony
!
voice channel 4/1/9 mode telephony
!
voice channel 4/1/10 mode telephony
!
voice channel 4/1/11 mode telephony
!
voice channel 4/1/12 mode telephony
!
voice destination VSD_1 h.323 address 195.167.10.33 1720
!
voice destination VSD_2 h.323 address 195.167.10.34 1720
!
voice destination to VSD_1 port 1 local channel VSD_1/1/1-24
!
voice destination to VSD_2 port 1 local channel VSD_1/1/1-24
!
voice phone group Ext. of PBX__1
!
voice phone group Ext. of PBX__2
!
```

voice phone group Ext. of PBX__1 type local extensions
!
voice phone group Ext. of PBX__2 type local extensions
!
voice phone group Ext. of PBX__1 site prefix off
!
voice phone group Ext. of PBX__2 site prefix off
!
voice phone group Ext. of PBX__1 format "xxxx"
!
voice phone group Ext of PBX_2 format "xxxx"
!
voice phone group Ext. of PBX__1 strip digit length 0
!
voice phone group Ext. of PBX__2 strip digit length 0
!
voice numbering plan to PBX_1
!
voice numbering plan to PBX_2
!
voice numbering plan to PBX_1 hunt method round robin
!
voice numbering plan to PBX_1 hunt method round robin
!
voice numbering plan to PBX_1 associate destination member to VSD_1
!
voice numbering plan to PBX_2 associate destination member to VSD_2
!
voice numbering plan to PBX_1 associate phone group member Ext. of PBX_1
!
voice numbering plan to PBX_2 associate phone group member Ext. of PBX_2
!
voice numbering plan to PBX_1 description trunk to route calls from VSD1 to PBX1
!
voice numbering plan to PBX_2 description trunk to route calls from VSD2 to PBX2
!
voice numbering plan to PBX_1
!
voice daughter card 4/1 activate
!

voice network h.323 out fast start

Command Usage

Set *outgoing* H.323 faststart mode for gateway (on/off).

Syntax Options

voice network {template *"TemplateName"* **| card** *slot/card_number* **} h.323 out[going] fast start {on | off}**

Definitions:

TemplateName	Identifies the voice network template by name, (e.g., **vsmvon1**); maximum length of 40 characters. Include quotes on each end of the name. The following characters are permitted in the voice network template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies the chassis slot number where VSM is installed, (e.g., **2**).
card_number	Specifies the voice daughtercard position number, (e.g., **1**).
going	Optional command syntax. You can type either **out** or **outgoing** in the command line.
on	Turns ON H.323 outgoing fast start for specified voice network template.
off	Turns OFF H.323 outgoing fast start for specified voice network template.

Default:
The default setting is **on**.

Command Examples:
voice network template salemvon1 h.323 outgoing fast start off
voice network template salemvon2 h.323 out fast start on
voice network card 2/1 h.323 outgoing fast start off
voice network card 2/2 h.323 out fast start on

Remarks

This command selects H.323 faststart mode on the outgoing side of the link. Faststart mode reduces the number of messages exchanged between endpoints.

H.323 faststart calls connect after a single round-trip message. The faststart information is attached to H.225 messages from general setup. H.225 setup messages contain information about voice channels proposed by the originator of the call. The terminating endpoint accepts one of the proposed channels, and informs the originator through the connect message. The connected endpoints then establish logical channels and switch to voice mode. Ringback is sent inband and the voice path exists when the remote endpoint picks up the phone. If the terminating endpoint picks up the phone before the voice channels are established, the originator receives the voice signal directly with no preceding ringback.

If faststart calls do not connect in the voice switching daughtercard, the switch automatically reverts the call to the general H.323 setup; this prevents fastart calls from being dropped.

H.225, which provides the call setup and control signaling needed to connect h.323 endpoints, is part of the H.323 signaling protocol stack. Q.931 is a similar protocol used over ISDN lines to set up, maintain and terminate calls between H.323 endpoints or agents.

Fastart commands take effect immediately and do not require use of the **voice daughter card activate** command.

voice daughter card h.323 in fast start

Command Usage

Set *incoming* H.323 faststart mode for gateway (on/off).

Syntax Options

voice daughter card <*slot/ card_number***> h.323 in[coming] fast start {on | off}**

Definitions:

slot	Specifies the chassis slot number where VSM is installed, (e.g., **2**).
card_number	Specifies the voice daughtercard position number, (e.g., **1**)
coming	Optional command syntax. You can type either **in** or **incoming** in the command line.
on	Turns ON H.323 incoming fast start for specified voice network template.
off	Turns OFF H.323 incoming fast start for specified voice network template.

Default:
The default setting is **on**.

Command Examples:
voice daughter card 2/1 h.323 incoming fast start off
voice daughter card 2/1 h.323 in fast start on

Remarks

For a brief description of faststart mode, see the **voice network h.323 outgoing faststart** command.

Fastart commands take effect immediately and do not require use of the **voice daughter card activate** command.

voice daughter card h.323 auto answer

Command Usage

Set automatic answering for incoming calls on gateway (on/off).

Syntax Options

voice daughter card <*slot/card_number*> h.323 auto[matic] answer {on | off}

Definitions:

slot	Specifies slot number of switching module installed in chassis, (e.g., **2**).
card_number	Specifies the voice daughtercard position number, (e.g., **1**).
matic	Optional command syntax. You can type either **auto** or **automatic** in the command line.
on	Turns ON automatic answer for incoming calls on gateway.
off	Turns OFF automatic answer for incoming calls on gateway.

Default:
The default setting is **off**.

Command Examples:
voice daughter card 2/1 h.323 automatic answer off
voice daughter card 2/2 h.323 auto answer on

Remarks

This command is used to select H.323 automatic call answering mode on the incoming side of the link; if turned ON, the call is connected automatically for instant voice.

voice daughter card first digit wait duration

Command Usage

Specify maximum time for voice daughtercard dialing timers to wait between off-hook state and first dialed tone (digit) to be detected.

Syntax Options

voice daughter card <*slot/card_number*> **first digit wait duration** <*timer_value*>

Definitions:

slot	Specifies slot number of switching module installed in chassis, (e.g., **2**).
card_number	Specifies the voice daughtercard position number, (e.g., **1**).
timer_value	Specifies maximum time for dialing timers to wait *between off-hook/first digit dialed* to be detected in milliseconds from 1 to 4,294,967,295 ms, (e.g., **10000**); (1 ms = 1/1000th of a second), (1,000 ms = 1 second), (10,000 ms = 10 seconds), etc. Refer to conversion table below to quickly determine the proper setting.

♦ **Syntax Note** ♦

Do not use commas when entering a dialing timer value (for example, **10,000** (10 seconds) will return a syntax error message).

Default:
The default *timer_value* is **10000**.

Command Examples:
voice daughter card 2/1 first digit wait duration 10000
voice daughter card 2/2 first digit wait duration 60000

Remarks

When this timer expires, a "no digits received" error condition occurs and the numbering plan dial attempt fails.

Use this table to quickly determine duration values for dialing timers.

Voice Switching Daughtercard Digit Durations (ms) Conversion Table			
10,000 ms = 10 seconds	**15,000 ms** = 15 seconds	**20,000 ms** = 20 seconds	**30,000 ms** = 30 seconds
60,000 ms = 60 sec. (1 min.)	**600,000 ms** = 10 min.	**900,000 ms** = 15 min.	**1,200,000 ms** = 20 min.
1,800,000 ms = 30 min.	**3,600,000 ms** = 60 min. (1 hour)	**21,600,000 ms** = 6 hours	**36,000,000 ms** = 10 hours
43,200,000 ms = 12 hours	**86,400,000 ms** = 24 hours (1 day)	**604,800,000 ms** = 7 days (1 week)	**864,000,000 ms** = 10 days
1,209,600,000 ms = 14 days (2 weeks)	**1,814,400,000 ms** = 21 days (3 weeks)	**2,419,200,000 ms** = 28 days (4 weeks)	**2,505,600,000 ms** = 29 days‘

Voice Switching Daughtercard Digit Durations (ms) Conversion Table			
2,592,000,000 ms = 30 days (1 month)	**2,678,400,000 ms** = 31 days	**4,233,600,000 ms** = 49 days	**4,294,967,295 ms** = 49 days, 17 hours, 2 min., 47 sec. and 295 ms

voice daughter card inter digit wait duration

Command Usage

Specify maximum time for voice daughtercard dialing timers to wait between tones (digits) being dialed.

Syntax Options

voice daughter card <*slot/card_number*> **inter digit wait duration** <*timer_value*>

Definitions:

slot	Specifies slot number of switching module installed in chassis, (e.g., **2**).
card_number	Specifies the voice daughtercard position number, (e.g., **1**).
timer_value	Specifies maximum time for dialing timers to wait *between digits being dialed* in milliseconds from 1 to 4,294,967,295 ms, (e.g., **5000**); (100 ms = 1/10 of a second), (1,000 ms = 1 second), (10,000 ms = 10 seconds), (60,000 ms = 1 minute), (300,000 = 5 minutes), etc. Refer to conversion table (see **voice daughter card first digit wait duration** command) to quickly determine the proper setting.

♦ **Syntax Note** ♦

Do not use commas when entering a dialing timer value (for example, **5,000** (5 seconds) will return a syntax error message).

Default:
The default *timer_value* is **5000**.

Command Examples:
voice daughter card 2/1 inter digit wait duration 5000
voice daughter card 2/2 inter digit wait duration 60000

Remarks

When this timer expires, unless a termination digit is dialed, it assumes the caller is finished dialing digits. The numbering plan in use then attempts a match.

voice daughter card dial time wait duration

Command Usage

Specify maximum time for dialing timers to wait for all tones (digits) to be dialed.

Syntax Options

voice daughter card <*slot/card_number*> **dial time wait duration** <*timer_value*>

Definitions:

slot — Specifies slot number of switching module installed in chassis, (e.g., **2**).

card_number — Specifies the voice daughtercard position number, (e.g., **1**).

timer_value — Specifies maximum time for dialing timers to wait *for all digits to be dialed*, from 1 to 4,294,967,295 ms, (e.g., **30000**); (100 ms = 1/10 of a second), (1,000 ms = 1 second), (10,000 ms = 10 seconds), (60,000 ms = 1 minute), (300,000 = 5 minutes), etc. second), (1,000 ms = 1 second), (10,000 ms = 10 seconds), (60,000 ms = 1 minute), (300,000 = 5 minutes), etc. Refer to conversion table (see **voice daughtercard first digit wait duration** command) to quickly determine the proper setting.

♦ **Syntax Note** ♦

Do not use commas when entering a dialing timer value (for example, **30,000** (30 seconds) will return a syntax error message).

Default:
The default *timer_value* is **120000**.

Command Examples:
voice daughter card 2/1 dial time wait duration 30000
voice daughter card 2/2 dial time wait duration 60000

Remarks

When this timer expires, unless a termination digit is dialed, it assumes the caller is finished dialing digits. The numbering plan in use then attempts a match.

voice daughter card termination digit

Command Usage

Specify digit used by voice daughtercard dialing timers to terminate dial process.

Syntax Options

voice daughter card <*slot/card_number*> **[no] termination digit** <*character*>

Definitions:

slot	Specifies slot number of switching module installed in chassis, (e.g., **2**).
card_number	Specifies voice daughtercard position number, (e.g., **1**).
no	Restores the digit to the default value **#**.
value	Specifies one of 16 characters: **1, 2, 3, 4, 5, 6, 7, 8, 9, 0, #, *, a, b, c, d,** or **no termination digit** (no commas allowed) that can be used by dialing timers to terminate dial process, (e.g., **#**). Dial termination digit cannot be part of valid phone number. (**a, b, c, d** values *not available this release.*)

♦ Syntax Note ♦

If a termination digit is used to terminate the dial process, only one digit (or single character) can be used for the value.

Default:
The default *character* value is **#**.

Command Examples:
voice daughter card 2/1 no termination digit
voice daughter card 2/2 termination digit 0
voice daughter card 2/3 termination digit #

Remarks

The **voice daughter card termination digit** command is used to determine when the dial process is complete. All other digits are ignored after the termination digit. The numbering plan in use attempts a match before the terminating digit is received. All digits dialed until the termination digit is received are considered valid, and the termination digit is discarded. Use of the termination digit is optional.

This command effects the behavior of all phone groups on the daughtercard.

voice port interface type

Command Usage

Specify voice daughtercard port *digital* connection interface type (does not include analog interface). This setting determines the number of channels per port.

Syntax Options

voice port <*slot/port*> interface type { t1 | e1 | e1 isdn pri | bri euro}

Definitions:

slot	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
t1	Specifies T1 as the voice daughtercard port connection interface type.
e1	Specifies E1 (QSIG) as the voice daughtercard port connection interface type.
e1 isdn pri	Specifies PRI E1 (ISDN) as the voice daughtercard port connection interface type.
bri euro	Specifies BRI Euro (Euro ISDN) as the voice daughtercard port digital connection interface type.

♦ **Syntax Notes** ♦

If PRI E1 is specified, this command sets the ISDN DS-1 type to PRI E1 (also referred to as ISDN).

If T1 is specified, Mu-law companding must be selected via the **voice signaling companding** command. Likewise, if E1 (either ISDN PRI or BRI) is specified, A-law companding must be selected.

When a digital voice port interface type is specified, a corresponding protocol type must also be specified via the **voice signaling protocol** command.

Default:
For *VSD only*, the default setting is **t1**. For *VSB only*, the default setting is **bri euro**.

Command Examples:
voice port 2/1 interface type t1
voice port 2/2 interface type e1
voice port 2/3 interface type e1 isdn pri
voice port 2/4 interface type bri euro

Remarks

The **voice port interface type** command is used to determine the number of channels allowed per physical port on the voice daughtercard; the interfaces as described below are only supported on the digital versions of this card. (The analog voice switching daughtercard (VSA) does not support any of these interfaces).

The T1 interface uses a maximum of 24 DS-0 (64 Kpbs) channels, has a capacity of 1.544 Mbps, and follows Mu-law companding, which is used in North America (United States and Canada), and Japan.

The E1 (QSIG/ISDN signaling) interface uses a maximum of 32 DS-0 (64 Kbps) channels, has a capacity of 2.048 Mbps, and follows A-law companding which is used in Europe. This interface is a European CEPT (Conference of European Postal and Telecommunications Administrations) carrier. Thirty channels can be used for voice. E1 uses timeslots rather than robbed-bit signaling. Timeslot 0 is for framing (synchronization) and timeslot 16 is for signaling (either Common Channel Signaling (CCS), i.e., (QSIG/ISDN), or Channel Associated Signaling (CAS)/R2, handshaking on multi-frequency signaling digit). Timeslot 1-15 and 16-31 are for voice channels.

The E1 ISDN PRI interface uses a maximum of 32 DS-0 (64 Kbps) channels, and has a capacity of 2.048 Mbps. The interface follows A-law companding and is a European CEPT carrier. Two channels are D (data) channels. Channel 16 is reserved as a control channel. Channel 0 is reserved for framing. Remaining channels are B (bearer) channels.

The BRI E1 (Euro ISDN) interface uses a maximum of 3 DS-0 (64 Kbps) channels, and has a capacity of 2.048 Mbps. The interface follows A-law companding and is a European CEPT carrier. Two voice channels are B (B1 and B2 bearer) channels supported at 64 Kbps; another channel supports data at 16 Kbps. Channel 0 is reserved as a control channel. This interface is only supported on the BRI voice switching digital daughtercard (VSB).

voice port frame format

Command Usage

Specifies the frame format of the voice port.

Syntax Options

voice port <*slot/port*> frame format {none | superframe | extended superframe | e1 | e1 crc | e1 mf | e1 crc mf}

Definitions:

slot	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
none	Specifies unframed voice port frame format.
superframe	Specifies superframe voice port frame format (also known as AT&T D4 format DS-1).
extended superframe	Specifies extended superframe (ESF) voice port frame format (DS-1 standard used for Wide Area Networks).
e1	Specifies CCITT (ITU Geneva) Recommendation G.704 (ITU-T Recommendation for synchronous frame structures used at primary and secondary levels; double frame (FAS / Pulse Code Modulation (PCM) 31) voice port frame format. CCITT stands for Consultive Committee on International Telegraphy and Telephony.
e1 crc	Specifies Cyclic Redundancy Check for transmission; CCITT Recommendation G.704; CRC4 multiframe (FAS/PCM 31).
e1 mf	Specifies G.704 with TS16 multiframing enabled; CRC4 double frame (MFAS/PCM 30). *(Not available this release.)*
e1 crc mf	Specifies G.704 with TS16 multiframing enabled; CRC4 multiframe (MFAS/PCM 30). *(Not available this release.)*

♦ **Syntax Notes** ♦

A frame format **none** means the frame is unframed.

If the voice daughtercard connection port type is set to T1, then only superframe, extended superframe and [no] are allowed. If the voice daughtercard connection port type is set to E1, E1 ISDN PRI, or BRI Euro, then only E1, E1 CRC, E1 MF, E1 CRC MF, and [no] are allowed. The port connection type can be set via the **voice port interface type** command.

A no frame format superframe, or no frame format extended superframe, or no frame format E1, or no frame format E1 CRC, or no frame format E1 MF, or no frame format E1 CRC MF means the frame is unframed; in which case, superframe, extended superframe, E1, E1 CRC, E1 MF, E1 CRC MF are ignored.

Default:

The default frame format setting is **none**.

Command Examples:

voice port 2/1 frame format none
voice port 2/2 frame format superframe
voice port 2/3 frame format extended superframe

Remarks

The **voice port frame format** command is used to indicate the type of DSL line implementing the circuit. The circuit affects the number of bits per second that the circuit can reasonably carry, as well as the interpretation of the usage and error statistics.

The time slot divisions, which are the basis for T1 and E1 circuit connections, e.g., multiplexed Digital Service (DS-1), are determined as follows for frames, superframes, extended superframes and multiframes.

T1 Framing

A T1 frame consists of 24, 8-bit time slots and a 1 bit-synchronization and control bit. Twelve (12) T1 frames can be grouped into a superframe (SF/D4), or 24 T1 frames can be grouped into an extended superframe. In each superframe, the 6th and 12th frame may contain "robbed bit" (A,B) signaling, which means the least significant bit is robbed from each time in the 6th and 12th frame and used for signaling. In extended superframes, this robbed bit signaling (A, B, C, D) occurs in the 6th, 12th, 18th, and 24th frames.

E1 Framing

The E1 frame consists of 32, 8-bit time slots (two of these slots are used for synchronization and multiframe signaling) for 256 bits per frame at 2.048 megabits per second. Sixteen (16) E1 frames are grouped into multiframe. An E1 multiframe can use Channel Associated Signaling (CAS) contained in time slot 16. Timeslot 16 in multiframe 0 is used for multiframe synchronization and control. Timeslot 16 of multiframes 1 through 15 are used to carry A, B, C, and D signaling bits.

voice port circuit identifier

Command Usage

Define voice port circuit identifier.

Syntax Options

voice port <*slot/port*> **circuit identifier** {*text_string*}

Definitions:

slot — Specifies slot number of voice switching daughtercard installed in switching module, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

text_string — Identifies vendor transmission circuit for troubleshooting, (e.g., **38.ivbd.005719.001.pt**); can be a maximum of 30 characters. The following characters are permitted in the text string: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ S % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Note** ♦

Circuit IDs tend to use this baseline format: "xx.xxx.xxxxxx.xxx"

At least one ASCII character must be used in the text string, and quotes must be located at each end of the circuit identifier.

Default:

None

Command Examples:

voice port 2/1 circuit identifier "38.ivbd.oo5719.001.pt"
voice port 2/2 circuit identifier "39.hqac.001727.000"
voice port 2/3 circuit identifier "37.hsgc.305001.508"

Remarks

The **voice port circuit identifier** command is used to identify transmission circuits (trunks) for troubleshooting by a telephone company. The circuit identifier is obtained from the telephone company who may need or require this identifying information in order to fix line transmission problems. Identification of the voice port circuit identifier is optional.

voice port nfas framing

Command Usage

Set E1 voice port NFAS (Non-Facility Associated Framing); (enable/disable).

Syntax Options

voice port <*slot/port*> nfas framing {enable | disable}

Definitions:

slot	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
enable	Turns ON NFAS framing on E1 voice port.
disable	Turns OFF NFAS framing on E1 voice port.

Default:
The default setting is **disable.**

Command Examples:
voice port 2/1 nfas framing disable
voice port 2/1 nfas framing enable

Remarks

The **voice port nfas framing** command determines whether NFAS framing, e.g, out-of-band signaling, will be used with E1 or PRI E1.

When NFAS framing is *enabled*, the framing is based on bit 2 of Time Slot 0 NOT-NFAS.

NFAS framing can only be enabled if the voice daughtercard connection port type is set to E1, E1 ISDN PRI, or BRI Euro. The port connection type can be set via the **voice port interface type** command.

voice port line build out

Command Usage

Set voice port line haul (short haul/long haul).

Syntax Options

voice port <*slot/port*> line build out { short [haul] | long [haul]}

Definitions:

slot	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
short	Specifies short haul line build out.
haul	Optional command syntax. You can type either **short** or **short haul** in the command line.
long	Specifies long haul line build out.
haul	Optional command syntax. You can type either **long** or **long haul** in the command line.

Default:
The default setting is **short haul**.

Command Examples:
voice port 2/1 line build out short haul
voice port 2/2 line build out long haul
voice port 2/3 line build out short
voice port 2/4 line build out long

Remarks

Indicates line build out of this port. Only T1/E1 ports with Line Interface Unit (LIU) equipped can support Long Haul (LH); otherwise, only Short Haul (SH) is supported.

To use this command, the voice daughtercard connection port type must be set to T1, E1 or EI ISDN PRI, or BRI Euro via the **voice port interface type** command.

voice port line length

Command Usage

Specify T1 voice port line length.

Syntax Options

voice port <*slot/port*> [line build out] line length <*value*>

Definitions:

slot	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., **2**)
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
line build out	Optional command syntax.
value	Specifies T1 voice port line length in ranges from 0 to 200 meters, (e.g., **30**).

Default:
The default line length *value* is **30**.

Command Examples:
voice port 2/1 line build out line length 30
voice port 2/2 line length 30

Remarks

To use this command, the voice daughtercard connection type must be set to T1 via the **voice port interface type** command.

To use this command, the voice daughtercard connection type must be set to **short haul** via the **voice port line build out** command.

voice port attenuation

Command Usage

Specify T1 voice port attenuation.

Syntax Options

voice port <*slot/port*> [line build out] attenuation {0 | -7.5 | -15.0 | -22.5}

Definitions:

slot	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
line build out	Optional command syntax.
0	Specifies 0 decibels. The decibels value indicates attenuation (i.e., the allowed decrease in power signal).
-7.5	Specifies -7.5 decibels. The decibels value indicates attenuation (i.e., the allowed decrease in power signal).
-15.5	Specifies -15.5 decibels. The decibels value indicates attenuation (i.e., the allowed decrease in power signal).
-22.5	Specifies -22.5 decibels. The decibels value indicates attenuation (i.e., the allowed decrease in power signal).

Default:
The default attenuation setting is **0** decibels.

Command Examples:
voice port 2/1 line build out attenuation 0
voice port 2/2 line build out attenuation -7.5
voice port 2/3 line build out attenuation -15.0
voice port 2/4 line build out attenuation - 22.5
voice port 3/1 attenuation 0
voice port 3/2 attenuation -7.5
voice port 3/3 attenuation -15.0
voice port 3/4 attenuation -22.5

Remarks

To use this command, the voice daughtercard connection type must be set to T1 via the **voice port interface type** command.

To use this command, the line build out must be set to long haul via the **voice port line build out** command.

voice port cable type

Command Usage

Specify E1, E1 ISDN PRI, or BRI Euro voice port cable type.

Syntax Options

voice port <*slot/port*> [line build out] cable type {75 | 120}

Definitions:

slot	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
line build out	Optional command syntax.
75	Specifies 75 Ohms for type of cable connected to port.
120	Specifies 120 Ohms for type of cable connected to port.

Default:
The default cable type setting is **120** Ohms.

Command Examples:
voice port 2/1 line build out cable type 120
voice port 2/2 line build out cable type 75
voice port 2/1 cable type 120
voice port 2/2 cable type 75

Remarks

To use this command, the voice daughtercard connection type must be set to E1, E1 ISDN PRI, or BRI Euro via the **voice port interface type** command.

voice port line coding

Command Usage

Specify line coding of voice port.

Syntax Options

voice port <*slot/port*> [line build out] line coding {ami | b8zs | hdb3}

Definitions:

slot	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
line build out	Optional command syntax.
ami	Specifies Alternative Mark Inversion line coding for the voice port.
b8zs	Specifies Bipolar 8 Zero substitution line coding for the voice port.
hdb3	Specifies High density Bipolar with 3 zero substitution line coding for the voice port.

Default:
The default line coding type is **ami**.

Command Examples:
voice port 2/1 line build out line coding ami
voice port 2/2 line build out line coding b8zs
voice port 2/3 line build out line coding hdb3
voice port 2/4 line coding ami
voice port 3/1 line coding b8zs
voice port 3/2 line coding hdb3

Remarks

AMI line coding is supported when the voice daughtercard connection port interface type is set to T1, E1, E1 ISDN PRI, or BRI Euro via the **voice port interface type** command.

The term dsx1AMI refers to a mode wherein no zero code suppression is present is used on the link because line encoding does not solve the problem directly. In this application, the higher layer must provide data which meets or exceeds the requirements, such as inverting High Level Data Link Control (HDLC) data. E1 links, with or without a Cyclic Redundancy Check (CRC), use dsx1AMI line coding.

B8Zs line coding is supported when the voice daughtercard connection port type is T1. The term dsx1B8ZS refers to the use of a specified pattern of normal bits and bipolar violations which are used to replace a sequence of 8 zero bits.

HDB3 line coding is supported when the voice daughtercard connection port type is E1, ISDN PRI E1, or BRI Euro.

voice port facilities data link protocol

Command Usage

Specify T1 voice port facilities data link protocol.

Syntax Options

voice port <*slot/port* > **facilities data link protocol {none | ansi t1.403 | at&t 54016 | t1.403 at&t}**

Definitions:

slot	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
none	Indicates device does not use the facilities data link.
ansi t1.403	Indicates device uses the dsx1ANSI-T1-403 facilities data link exchange recommended by ANSI.
at&t 54016	Indicates device uses the dsx1ATT-54016 ESF (Extended Super Frame) facilities data link exchange.
t1.403 at&t	Indicates device uses the ANSI t1.403 for ESF (Extended Super Frame) facilities data link exchange.

♦ **Syntax Notes** ♦

A "no facilities data link protocol" means that the frame is unframed.

A "no facilities data link protocol ANSI T1.403," or "no facilities data link protocol AT&T 54016," or "no facilities data link protocol T.1403 AT&T," means that the device does not use the facilities data link protocol. The "ANSI T1.403 and AT&T 54106 and T1.403 AT&T are ignored.

Default:
The default setting is **none**.

Command Examples:
voice port 2/1 facilities data link protocol none
voice port 2/2 facilities data link protocol ansi t1.403
voice port 2/3 facilities data link protocol at&t 54016
voice port 2/4 facilities data link protocol t1.403 at&t

Remarks

To use this command, the voice daughtercard connection interface type must be set to T1 via the **voice port interface type** command.

voice port facilities data link port role

Command Usage

Set T1 voice port facilities data link port role (network/user).

Syntax Options

voice port <*slot/port*> facilities data link port role {network | user}

Definitions:

slot	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
network	Indicates facilities data link port is controlled by the network, i.e., the VSD.
user	Indicates facilities data link port is controlled by the user, e.g., the telephone company.

♦ **Syntax Note** ♦

If the port role is network and the **fdlMode** is set to AT&T 54016 via the **voice port facilities data link protocol** command, then this port periodically sends AT&T performance requests to customer interface.

Default:
The default setting is **user**.

Command Examples:
voice port 2/1 facilities data link port role user
voice port 2/2 facilities data link port role network

Remarks

To use this command, the voice daughtercard connection interface type must be set to T1 via the **voice port interface type** command.

voice port transmit clock source

Command Usage

Specify voice port transmit clock source.

Syntax Options

voice port <*slot/port*> transmit clock source {loop timing | local timing}

Definitions:

slot	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
loop timing	Indicates that the recovered receive clock is being used as the transmit clock.
local timing	Indicates that a local clock source is being used as the transmit clock.

Default:
The default setting is **local timing**.

Command Examples:
voice port 2/1 transmit clock source local timing
voice port 2/2 transmit clock source loop timing

voice port loop back mode

Command Usage

Specify T1 voice port loop back mode configuration.

Syntax Options

voice port <*slot/port*> loop back mode {none | payload | line | inward}

Definitions:

slot	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
none	Indicates T1 interface sends looped or normal data for loopback (i.e., not in the loopback state; a device incapable of performing a loopback on the interface always returns "none" as its value. *Also known as Dsx1NoLoop.*
payload	Indicates T1 interface sends request for a payload loopback (i.e., received signal at this interface is looped through the device; typically, the received signal is looped back for retransmission after it has passed through the device's framing function). *Also known as Dsx1PayloadLoop.*
line	Indicates T1 interface sends request for a line loopback (i.e., received signal at this interface does not go through the device (minimum penetration) but is looped back out). *Also known as Dsx1LineLoop.*
inward	Indicates undefined T1 interface loopback request. *Also known as Dsx1OtherLoop.*

Default:
The default setting is **payload.**

Command Examples:
voice port 2/1 loop back mode none
voice port 2/2 loop back mode payload
voice port 2/3 loop back mode line
voice port 2/4 loop back mode inward

Remarks

This variable represents the loop back configuration of the T1 (DS-1) interface, and indicates what type of code is being sent across the T1 interface by the device. A bad value is returned in response to a requested loop back state that the interface providing read/write access does not support.

voice channel signaling mode

Command Usage

Specify voice port channel signaling mode.

Syntax Options

voice channel <*slot/port*> **signaling mode {none | cas | ccs}**

Definitions:

slot	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
none	Indicates that no bits are reserved for signaling on this channel.
cas	Indicates that T1 Channel Associated Signaling (CAS) is in use; (applies only tó VSD T1).
ccs	Indicates that Common Channel Signaling (CCS) is in use on channel 16 of an E1 link (applies to VSD E1, VSD E1 ISDN PRI), and to channel 3 (VSB BRI Euro voice daughtercard).

Default:
Refer to definitions above.

Command Examples:
voice port 2/1 signaling mode none
voice port 2/2 signaling mode CAS
voice port 2/3 signaling mode CCS

Remarks

A “no signal mode” means that no bits are reserved for signaling on this channel.

A “no signal mode CAS” or “no signal mode CCS” means that no bits are reserved for signaling on this channel. The CAS and CCS are ignored.

voice port trap generation

Command Usage

Set voice port trap generation (enable/disable). Indicates if line status change trap is sent to the network management system (NMS).

Syntax Options

voice port <*slot/port*> trap generation {enable | disable}

Definitions:

slot	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
enable	Turns ON trap generation on voice port.
disable	Turns OFF trap generation on voice port.

Default:
The default setting is **disable.**

Command Examples:
voice port 2/1 trap generation disable
voice port 2/2 trap generation enable

Remarks

To use this command, the voice daughtercard connection type must be set to T1, E1, E1 ISDN PRI, or BRI Euro via the **voice port interface type** command.

voice port send code

Command Usage

Specify T1 voice port loop back send configuration values. (*Not available this release.*)

Syntax Options

voice port <*slot/port*> send code {none | line | payload | reset | quasi [random signal] | 511 [pattern] | 3 in 24 [pattern] | other}

Definitions:

slot	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
none	Indicates T1 interface sends looped or normal data for loopback (i.e., not in the loopback state; a device incapable of performing a loopback on the interface always returns "none" as it's value. *Also known as dsx1SendNoCode.*
line	Indicates T1 interface sends request for a line loopback (i.e., received signal at this interface does not go through the device (minimum penetration) but is looped back out). *Also known as dsx1SendLineCode.*
payload	Indicates T1 interface sends request for a payload loopback (i.e., received signal at this interface is looped through the device; typically, the received signal is looped back for retransmission after it has passed through the device's framing function). *Also known as dsx1SendPayloadCode.*
reset	Indicates undefined T1 interface loopback termination request. *Also known as dsx1SendResetCode.*
quasi	Indicates T1 interface sends Quasi-Random Signal test. *Also known as dsx1SendQRS.*
random signal	Optional command syntax.
511	Indicates T1 interface sends 511 bit fixed test pattern. *Also known as dsx1Send511Pattern.*
pattern	Optional command syntax.
3 in 24	Indicates T1 interface sends fixed text pattern of 3 bits set in 24. *Also known as dsx1Send3in24Pattern.*
other	Indicates T1 interface sends undefined test pattern. *Also known as dsx1SendOtherTestPattern.*

Default:
The default setting is **payload**.

Command Examples:
voice port 2/1 send code none
voice port 2/2 send code line
voice port 2/3 send code payload
voice port 2/4 send code reset
voice port 3/1 send code quasi random signal
voice port 3/2 send code quasi
voice port 3/3 send code 511 pattern
voice port 3/4 send code 511
voice port 4/1 send code 3 in 24 pattern

Remarks

This variable represents the loop back configuration of the T1 (DS-1) interface, and indicates what type of code is being sent across the T1 interface by the device. A bad value is returned in response to a requested loop back state that the interface providing read/write access does not support.

voice port isdn connection protocol

Command Usage

Specify ISDN connection protocol for ISDN (BRI Euro) ports.

Syntax Options

voice port <*slot/port*> isdn connection protocol {net | user | qmaster | qslave}

Definitions:

slot	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
net	Sets the ISDN connection protocol to network mode for PRI E1 (European Telecommunications Standards Institute (ETSI) standard protocol).
user	Sets the ISDN connection protocol to user mode for PRI E1 (ETSI standard protocol).
qmaster	Sets the QSIG ISDN connection protocol to QSIG standard protocol for PRI E1; QSIG *master* (ETSI std. protocol).
qslave	Sets the ISDN connection protocol to QSIG standard protocol for PRI E1; QSIG *slave* (ETSI std. protocol).

Default:
The default setting is **qmaster**.

Command Examples:
voice port 2/1 isdn connection protocol qmaster
voice port 2/2 isdn connection protocol net
voice port 2/3 isdn connection protocol user
voice port 2/4 isdn connection protocol qslave

Remarks

The **voice port isdn connection protocol** command is used to configure the channel protocol mode for the ISDN ports at the voice daughtercard level. Specification of the protocol is optional.

To used this command, the voice daughtercard digital connection port type must be set to BRI Euro via the **voice port interface type** command.

voice port isdn switch type

Command Usage

Specify ISDN connection switch type for ISDN ports.

Syntax Options

voice port <*slot/port*> isdn switch type {net3 | net5}

Definitions:

slot	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
net3	Sets ISDN connection switch type to Euro ISDN (BRI E1 2 B + D channels) *(Not available this release.)*
net5	Sets ISDN connection switch type to Euro ISDN (PRI E1 30 B + D channels).

♦ **Syntax Notes** ♦

If the voice daughtercard connection port type is set to PRI E1, the **voice port isdn connection switch type** command uses the default value of **net5**.

The port connection type must be set via the **voice port interface type** command to indicate the number of channels available on the port, e.g., 3 channels (BRI Euro), 32 channels (E1 PRI ISDN).

Default:
The default setting is **net5**.

Command Examples:
voice port 2/1 isdn switch type net5
voice port 2/2 isdn switch type net3

Remarks

The **voice port isdn connection switch type** command is used to configure the type of switch connection for the ISDN ports at the voice daughtercard level. Specification of the E1 ISDN protocol is optional unless E1 ISDN PRI or BRI Euro is used.

In the *"user mode,"* set through the **voice port isdn connection protocol** command, the ISDN connection switch type uses the switch type to which the ISDN link is connected.

In the *"network mode,"* set through the **voice port isdn connection protocol** command, the ISDN connection switch type commands selects the switch type to be emulated.

voice channel isdn d channel

Command Usage

Specify control (data or "D") channels for E1 ISDN PRI or BRI Euro.

Syntax Options

voice channel <*slot/port/channel*> **isdn d channel**

Definitions:

slot	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
channel	Specifies the control (D) channel number (i.e., **16** for E1 ISDN PRI and **0** for BRI Euro); BRI Euro *not available this release.*

Syntax Notes ♦

The port connection type must also be set via the **voice port interface type** command to indicate the number of channels available on the port, e.g., 3 channels (BRI Euro), 32 channels (E1 PRI ISDN).

Only **0** and **16** are allowed for data channel numbers; control channel **16** is reserved for signaling on E1 ISDN PRI, and **0** is reserved for the BRI Euro control channel. For E1 ISDN PRI, channel **0** is used for framing (there is no framing on BRI Euro).

Data and corresponding bearer channels must apply to the same voice switching daughtercard.

Default:

For *E1 ISDN PRI*, the default *channel* value is **16**. For *BRI Euro*, the default *channel* value is **0**.

Command Example:

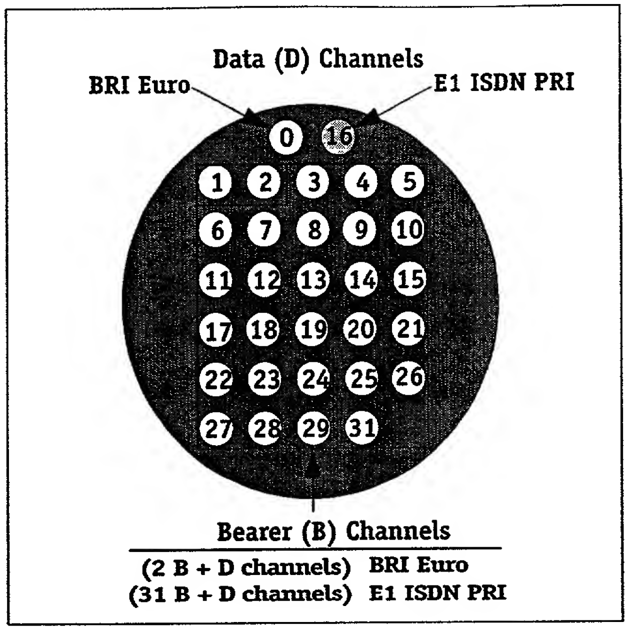
voice channel 2/1 16 isdn d channel
voice channel 2/2 16 isdn d channel
voice channel 2/3 16 isdn d channel
voice channel 2/4 16 isdn d channel
voice channel 2/1 0 isdn d channel
voice channel 2/2 0 isdn d channel
voice channel 2/3 0 isdn d channel
voice channel 2/4 0 isdn d channel

Remarks

The **voice channel isdn d channel** command is used to configure the type of switch connection for the ISDN ports at the voice daughtercard level. Specification of the ISDN protocol is optional if not using E1 ISDN PRI or BRI Euro.

Use this table and illustration below to determine valid data/bearer channel, slot and port values (applies to OS/R with two voice switching daughtercards).

Interface Type	Data Channels	Bearer Channels	Slot Number	Port Number
E1 ISDN PRI	16	1-15, 17-31	2-9	1-4
BRI Euro	0	1-2	2-9	1-4


Data (D) Channels
BRI Euro
E1 ISDN PRI
0
16
1 2 3 4 5
6 7 8 9 10
11 12 13 14 15
17 18 19 20 21
22 23 24 25 26
27 28 29 31
Bearer (B) Channels
(2 B + D channels) BRI Euro
(31 B + D channels) E1 ISDN PRI

voice channel isdn b channel

Command Usage

Specify E1 ISDN bearer ("B") channels. (*Not available this release.*)

Syntax Options

voice channel <*slot/port/channel*> **isdn b channel** <*port/dchannel dsl_id*>

<u>Definitions</u>:

slot	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
channel	Specifies bearer (B) channel number (e.g., **1, 2, 3, 4, 5, 6, 7, 8, 9, 10, 11, 12, 13, 14, 15, 17, 18, 19, 20, 21, 22, 23, 24, 25, 26, 27, 28, 29, 31**) for E1 ISDN PRI; BRI Euro is limited to two bearer channels.
port	
dchannel	
dsl_id	

♦ Syntax Notes ♦

The port connection type must also be set via the **voice port interface type** command to indicate the number of channels availalble on the port, e.g., 3 channels (BRI Euro), 32 channels (E1 PRI ISDN).

Only **0** and **16** are allowed for data channel numbers; control channel **16** is reserved for signaling on E1 ISDN PRI, and **0** is reserved for the BRI Euro control channel. For E1 ISDN PRI, channel **0** is used for framing (there is no framing on BRI Euro).

Bearer channels and corresponding Data channels must apply to to the same voice switching daughtercard.

<u>Default</u>:
For *E1 ISDN PRI*, the default *channel* values are **1-15, 17-31**. For *BRI Euro*, the default *channel* values are is **1-2**.

<u>Command Example</u>:
voice channel 2/1 1 isdn b channel
voice channel 2/2 2 isdn b channel
voice channel 3/1 1 isdn b channel
voice channel 3/2 15 isdn b channel
voice channel 3/3 17 isdn b channel
voice channel 3/4 31 isdn b channel

Remarks

The **voice port isdn connection switch type** command is used to configure the type of switch connection for the ISDN ports at the voice daughtercard level. Specification of the ISDN protocol is optional if not using E1 ISDN PRI or BRI Euro.

Use the table and illustration above to determine valid bearer/data channel, slot and port values (applies to OS/R with two voice switching daughtercards).

Channel Properties

The commands listed and described below are used to configure the channel properties for individual voice channels as follows: voice channel mode, PLAR (Private Line Automatic Ring-down), outbound caller ID, and voice channel initialization.

Voice Channel Configuration

voice channel mode

voice channel PLAR dial-in phone number

Channel Operational State

voice channel initialization (in-service/out-of-service)

voice channel mode

Command Usage

Specify voice channel mode—telephony, passthrough or PLAR (Private Line Automatic Ringdown) used to complete calls on designated channels. Establishes semi-fixed bandwidth connections between points in the network. (*Not available this release.*)

Syntax Options

voice channel <*slot/port/startChannel-endChannel*> **mode {telephony | pass through | plar}**

Definitions:

slot	Specifies slot number of voice daughtercard installed in switching module, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **2**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦
Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

telephony	Allows use of Numbering Plan to complete calls to specified hunted destination.
pass through	Allows two DS-0 channels to be cross-connected to complete calls (*not available this release*)
plar	Allows calls to be routed based upon telephone number configured to dialing plan.

Default:
The default setting is **telephony**.

Command Examples:
voice channel 2/2/1-12 mode telephony
voice channel 2/2/13-24 mode pass through
voice channel 2/2/1-12 mode plar

Remarks

In the *telephony mode*, digits come from voice path or signaling. Upon in-seize, the voice switching daughtercard collects inbound digits, then uses the Numbering Plan to complete calls to the specified hunted destination.

In the *passthrough mode*, no digit processing is performed by the DSPs on the voice switching daughtercard, and no routing is performed. Upon in-seize, the channel is immediately connected to another channel, via Numbering Plan processing.

In the *PLAR mode*, no inbound digit processing takes place. Upon in-seize, the PLAR telephone number is immediately routed based upon the dialing plan. The telephone number must be configured to automatically dial using the dial plan.

voice channel dial in private line automatic ringdown

Command Usage

Specify voice channel Private Line Automatic Ringdown (PLAR) dial-in phone number. This command is used to configure the number to be dialed in Switched CAS mode when a specified channel goes off-hook.

Syntax Options

voice channel <*slot/port/startChannel-endChannel*> **dial in private line automatic ringdown** < *"plar phone number"* >

Definitions:

slot	Specifies slot number of voice daughtercard installed in switching module, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **2**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**). ♦ Syntax Note ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
plar phone number	Telephone number configured to dialing plan (e.g., **8188803500**). ♦ Syntax Notes ♦ If no number is entered for the PLAR dial-in phone number, operation will be as normal. This command must be issued before activating the voice daughtercard.

Default:
None

Command Examples:
voice channel 2/2/1-12 dial in private automatic line ringdown 8188803500
voice channel 2/3/13-24 dial in private automatic line ringdown 8188803501

Remarks

To use this command, the channel type (mode) must be set to PLAR (Private Line Automatic Ringdown) via the **voice channel mode** command.

This command is normally used for calls placed from courtesy phones.

voice channel state

Command Usage

Set voice channel initialization (in-service/out-of-service). Signifies the initial state (admin-status) of the channel, or attempt to modify the state of the channel.

Syntax Options

voice channel <*slot/port/startChannel-endChannel*> state {in service | out of service}

Definitions:

slot	Specifies slot number of voice daughtercard installed in switching module, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

in service	Puts channel in service as soon as possible after configuration.
out of service	Keeps channel out of service and unused. Channel does not accept hunting or in/out seizes.

Default:
The default setting is **out of service**.

Command Examples:
voice channel 2/1/1-12 state in service
voice channel 2/2/13-24 state out of service

Remarks

If the configuration of this channel is incomplete, the command is ignored, the channel is NOT placed into service, and error conditions result.

Telephony Signaling Template/Signaling Attributes

The commands listed and described below are used to configure Telephony Signaling templates and associated signaling attributes including general signaling, Ear & Mouth (E&M), Foreign Exchange Station (FXS), and Foreign Exchange Office (FXO) signaling attributes. This entails call signaling capabilities, outbound caller ID, call progress tones, echo and acoustic echo cancellers, and overrides for call signaling.

Telephony Signaling Template (create, delete or view)

Telephony Signaling Template Protocol

Telephony Signaling Template Channel (assign and view) (Assign *not available this release.*)

Signaling Attributes

Dial Out Signaling Tones

time to wait before first tone is sent

duration for a single tone

duration to pause between tones

out dialing port type

Channel Timing

maximum call time length

time to wait for call to be answered

time to wait to force caller to disconnect

time to wait to tear down fax call

Signal Power

companding type (Mu-law/A-law)

gain inserted at *receiver*

gain inserted at *transmitter*

amplitude of comfort (idle) noise

E&M Common Signaling

E&M signaling time for *transition to off-hook* (debounce)

E&M signaling time for *transition to on-hook* (debounce)

E&M signaling time to wait before *declaring on-hook* (seize detect)

E&M signaling time to wait before *declaring off-hook* (clear detect)

E&M signaling time to wait before confirming on-hook

E&M signaling time to wait for on-hook after a clear

E&M signaling time to wait between termination and *origination*

E&M signaling time to wait between termination and *receiving*

E&M signaling dial tone generation on incoming calls (on/off)

minimum E&M signaling connection time

time to wait after E&M signaling hang up before disabling

E&M Wink Start Signaling

minimum E&M wink delay on incoming calls

maximum E&M wink delay on incoming calls

duration of E&M wink on incoming calls

time to ignore tones after E&M wink

time to wait for E&M wink on outgoing calls

minimum E&M wink duration

maximum E&M wink duration

E&M Immediate Start Signaling

E&M immediate start time to remain off-hook when congested

E&M immediate start time to wait before beginning digit collection

E&M Delay Start Signaling

minimum E&M delay start response to off-hook state

maximum E&M delay start response to off-hook state

time to ignore incoming digits after E&M delay start

E&M delay start signal detection

minimum E&M delay start detection time on “M” lead

maximum E&M delay start detection time on “M” lead

maximum time to wait for E&M delay start detection

Foreign Exchange Station (FXS) Loop Start (LS) Signaling

FXS LS debounce interval to *on-hook* transition

FXS LS debounce interval to *off-hook* transition

FXS LS time to wait before declaring off-hook

FXS LS minimum time to wait before declaring on-hook by *originator*

FXS LS minimum time to wait before declaring on-hook by *answerer*

FXS LS time to wait after supervisory disconnect before declaring on-hook

FXS LS duration of supervisory disconnect

FXS LS to *generate outbound* caller ID (on/off)

FXS LS cadence coefficient (North America/Europe)

FXS LS ring ID

Foreign Exchange Office (FXO) Loop Start (LS) signaling

FXO LS incoming ring signal debounce interval

FXO LS debounce interval to on-hook transition

FXO LS *supervisory disconnect* detection signal (enable/disable)

FXO LS *duration of supervisory disconnect* detection signal

FXO LS time before originating calls while receiving calls

FXO LS time between ring *cycles* to detect ringing

FXO LS time between ring *pulses* to detect ringing

FXO LS to *detect inbound* caller ID (on/off)

FXO LS number of rings allowed before answering

FXO LS debounce for loop current detector

FXO LS debounce for battery reversal detector

Foreign Exchange Station (FXS) Ground Start (GS) Signaling

FXS GS time to wait before declaring off-hook

FXS GS debounce interval for on-hook transition

minimum FXS GS time to wait before declaring on-hook by *originator*

minimum FXS GS time to wait before declaring on-hook by *answerer*

FXS GS time to wait after ring ground before grounding tip

maximum FXS GS time to wait for loop to close after grounding tip

minimum FXS GS start time between open loop and idle state

FXS GS to *generate outbound* caller ID

FSX GS debounce interval for off-hook

FXS GS debounce interval for ring ground detector

FXS GS cadence coefficient (North America/Europe)

FXS GS ring ID

Foreign Exchange Station (FXO) Ground Start (GS) Signaling

FXO GS debounce interval for loop open detection

maximum FXO GS time between ring ground and tip ground

FXO GS debounce interval for tip ground detector

FXO GS debounce for incoming ring signal

FXO GS time between consecutive ring *cycles*

FXO GS time between consecutive ring *pulses*

FXO GS to *detect inbound* caller ID (on/off)

FXO GS number of rings allowed before answering

FXO GS debounce interval for loop current detector

FXO GS debounce interval for battery reversal detector

Outbound Caller ID

outbound caller ID name (private/unavailable) to transmit

outbound caller ID number (published/non-published) to transmit

Tones

outbound tone table (ringing/silence)

call progress tone detection (on/off/relative)

call progress tone detection configuration (default/alternate)

V.18 tone detection *threshold hang time*

V.18 tone detection *threshold level*

V.18 *single tone* detection threshold level

V.18 *single tone* detection threshold time

Echo Canceller

echo canceller non-linear sensitivity

Acoustic Echo Canceller

acoustic echo canceller mode (on/off)

acoustic echo canceller non-linear processor (on/off)

acoustic echo canceller output (on/off)

acoustic echo canceller handset (hs) speaker gain

acoustic echo canceller handsfree (hf) speaker gain

Override Call Signaling Capabilities

override call signaling for in-band call progress tones (on/off)

override call signaling for full call progress tones (on/off)

override call signaling for ring back (on/off)

override call signaling for in-band codec switching (on/off)

override call signaling for packet switch (PSU) codec switching (on/off)

override call signaling for network overlap dialing (on/off)

override call signaling for information element (IE) transport (on/off)

override call signaling for QSIG information (IE) transport (on/off)

override call signaling for voice, fax, modem, data setup (on/off)

voice signaling template

Command Usage

Create Telephony Signaling template with specified telephony interface name to uniquely identify the template.

Syntax Options

voice signaling template < *“TemplateName”* >

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }.

♦ **Syntax Notes** ♦

This command must be issued before any parameters can be added to the template.

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

Default:
None

Command Examples:
voice signaling template “sigtempbranch1”
voice signaling template “sigtempbranch2”
voice signaling template “sigtempbranch3”

Remarks

The Telephony Signaling template is used to configure the telephony signaling of the physical ports on the voice switching daughtercard. The templates are assigned to ports on a voice switching daughtercard via the **voice channel assign signaling template** command.

voice no signaling template

Command Usage

Delete Telephony Signaling template with specified telephony interface name to remove the template.

Syntax Options

voice no signaling template < *"TemplateName"* **>**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }.

♦ **Syntax Note** ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

Default:

None

Command Examples:

voice no signaling template "sigtempbranch1"
voice no signaling template "sigtempbranch2"
voice no signaling template "sigtempbranch3"

view voice signaling template

Command Usage

Display Telephony Signaling template with specified telephony interface name to view the template.

Syntax Options

view voice signaling template < *"TemplateName"* **>**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }.

♦ Syntax Note ♦
At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

Default:
None

Command Examples:
view voice signaling template "sigtempbranch1"
view voice signaling template "sigtempbranch2"
view voice signaling template "sigtempbranch3"

Screen Output

To view parameters for a voice signaling template, type **view voice signaling template** and a valid signaling template name, e.g., **view voice signaling template "emi with dialtone"** and then press **<Enter>**.

A screen similar to the following displays.

```
*******************************************
Viewing Signaling Template
*******************************************
!
voice signaling template EMI with dialtone
!
voice signaling template EMI with dialtone protocol emi
!
voice signaling template EMI with dialtone companding mulaw
!
voice signaling template EMI with dialtone em dial tone on
!
voice signaling template EMI with dialtone emi glare report 5500
!
voice signaling template EMI with dialtone emi digit wait 250
!
```

voice channel assign signaling template

Command Usage

Assign Telephony Signaling interface template to specified channel(s). (*Not available this release;* all commands using the syntax *"TemplateName"* are currently not applicable as a result).

Syntax Options

voice channel <*slot/port/startChannel-endChannel*> **assign signaling template** < *"TemplateName"* >

Definitions:

slot	Specifies chassis slot number where VSM is installed (e.g., **2**).
port	Specifies physical port number on voice daughtercard (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }.

♦ **Syntax Notes** ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

This command must be issued before the voice switching daughtercard can be activated.

If using ISDN (E1 ISDN PRI or BRI Euro) all channels on a port should be configured as ISDN in one instance of this assign command.

Default:
None

Command Examples:
voice channel 2/1/1-12 assign signaling template "sigtempbranch1"
voice channel 2/2/13-24 assign signaling template "sigtempbranch2"
voice channel 2/3/31-30 assign signaling template "sigtempbranch3"

Remarks

To use this command for ISDN, the voice daughtercard connection type must be set to E1 PRI ISDN or BRI Euro via the **voice port interface type** command.

If the Telephony Signaling template specifies use of the ISDN protocol, then all channels on the port that a template has been assigned must be set to ISDN (PRI E1).

view voice signaling channel

Command Usage

Display Telephony Signaling channel(s).

Syntax Options

view voice signaling channel <*slot/port/startChannel-endChannel*>

Definitions:

slot	Specifies chassis slot number where VSM is installed (e.g., **2**).
port	Specifies physical port number on voice daughtercard (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

Default:
None

Command Examples:
view voice signaling channel 2/1/1-12
view voice signaling channel 2/2/13-12
view voice signaling channel 2/3/31-30

Screen Output

To view parameters for a voice signaling channel, type **view voice signaling channel** and a valid voice signaling channel, e.g., **view voice signaling channel 4/1/1**, and then press **<Enter>**.

A screen similar to the following displays.

```
********************************************
Viewing Signaling Channel
********************************************
!
voice signaling channel 4/1/1 protocol emi
!
voice signaling channel 4/1/1 companding mulaw
!
voice signaling channel 4/1/1 em dial tone on
!
voice signaling channel 4/1/1 emi glare report 5500
!
voice signaling channel emi digit wait 250
!
```

voice signaling protocol

Command Usage

Specify the protocol to use with a Telephony Signaling template.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} protocol {fxs ls | fxo ls | fxs gs | fxo gs | emi | emd | emw | isdn | trans cas | trans ccs}**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }.

♦ **Syntax Notes** ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed (e.g., **2**).

port — Specifies physical port number on voice daughtercard (e.g., **1**).

fxs ls — Specifies (CAS) foreign exchange station loop start signaling as the signaling template protocol.

fxo ls — Specifies (CAS) foreign exchange office loop start as the signaling template protocol.

fxs gs — Specifies (CAS) foreign exchange station ground start as the signaling template protocol.

fxo gs — Specifies (CAS) foreign exchange office ground start as the signaling template protocol.

emi — Specifies (CAS) E&M (ear & mouth) immediate start as the signaling template protocol.

emd — Specifies (CAS) E&M (ear & mouth) delay start as the signaling template protocol.

emw — Specifies (CAS) E&M (ear & mouth) wink start as the signaling template protocol.

isdn — Specifies (CCS) Integrated Services Digital Network (ISDN) as the signaling template protocol.

trans cas — Specifies transparent mode channel associated signaling (CAS) as the signaling template protocol. *(Not available this release.)*

trans ccs — Specifies transparent mode common channel signaling (CCS) as the signaling template protocol. *(Not available this release.)*

Default:

The default setting is **fxs ls**.

Command Examples:

voice signaling template "sigtempbranch1" protocol fxs ls
voice signaling template "sigtempbranch2" protocol fxo ls
voice signaling template "sigtempbranch3" protocol fxs gs
voice signaling template "sigtempbranch4" protocol fxo gs
voice channel 2/1/1-12 protocol emi
voice channel 2/2/13-24 protocol emd
voice channel 2/3/1-30 protocol emw
voice channel 2/4/1-30protocol isdn

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

To use this command for ISDN, the voice daughtercard connection type must be set to ISDN PRI E1 via the **voice port interface type** command. If the Telephony Signaling template specifies use of the ISDN protocol, then all channels on the port that a template has been assigned must be set to ISDN (E1 ISDN PRI or BRI Euro); the setting must also match the daughtercard connection type set via the **voice port interface type** command.

voice signaling out wait

Command Usage

Specify time to wait before first tone (digit) is sent (dialed out) after going off-hook.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} out wait** <*value*>

Definitions:

TemplateName — Identifies the signaling template by name (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ` ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ Syntax Notes ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies slot number of voice daughtercard installed in switching module, (e.g., **2**).

port — Specifies physical port number on voice daughtercard (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ Syntax Note ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — When tone dialing is used, specifies in milliseconds from 0 ms to 20,000, the time to delay after going off-hook before sending the first outbound dial digit (e.g., **20000**).

♦ Syntax Note ♦

Do not use commas when entering the time to delay after going off-hook before sending the first outbound dial digit (for example, **20,000** will return a syntax error message).

Default:

The default value is **400**.

Command Examples:

voice signaling template "sigtempbranch1" outwait 0
voice signaling template "sigtempbranch2" outwait 400
voice signaling template "sigtempbranch3" outwait 20000
voice signaling channel 2/1/1-12 outwait 0
voice signaling channel 2/2/13-24 outwait 400
voice signaling channel 2/3/1-30 outwait 20000

voice signaling out tone digit duration

Command Usage

Specify duration for a single tone (digit) dialed out.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} out tone digit duration <***value***>**

Definitions:

TemplateName	Identifies the signaling template by name (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
	♦ **Syntax Notes** ♦ At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.
slot	Specifies slot number of voice daughtercard installed in switching module, (e.g., **2**).
port	Specifies physical port number on voice daughtercard (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).
	♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
value	When tone dialing is used, specifies in milliseconds from 0 ms to 2000, the duration of a tone, (e.g., **2000**).
	♦ **Syntax Note** ♦ *Do not* use commas when entering the duration of a tone (for example, **2,000** will return a syntax error message).

Default:
The default value is **200**.

Command Examples:
voice signaling template "sigtempbranch1" out tone digit duration 0
voice signaling template "sigtempbranch2" out tone digit duration 200
voice signaling template "sigtempbranch3" out tone digit duration 2000
voice signaling channel 2/1/1-12 out tone digit duration 0
voice signaling channel 2/2/13-24 out tone digit duration 200
voice signaling channel 2/3/1-30 out tone digit duration 2000

voice signaling out tone interdigit duration

Command Usage

Specify duration to pause between tones (digits) dialed out.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} out tone interdigit duration** <*value*>

Definitions:

TemplateName — Identifies the signaling template by name (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies slot number of voice daughtercard installed in switching module, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — When tone dialing is used, specifies in milliseconds from 0 ms to 2000, the duration of the interdigit gap between tones, (e.g., **200**).

♦ **Syntax Note** ♦

Do not use commas when entering the duration of the interdigit gap between tones (for example, **2,000** will return a syntax error message).

Default:
The default value is **200**.

Command Examples:

voice signaling template "sigtempbranch1" out tone interdigit duration 0
voice signaling template "sigtempbranch2" out tone interdigit duration 200
voice signaling template "sigtempbranch3" out tone interdigit duration 2000
voice signaling channel 2/1/1-12 out tone interdigit duration 0
voice signaling channel 2/2/13-24 out tone interdigit duration 200
voice signaling channel 2/3/31-30 out tone interdigit duration 2000

voice signaling out dialing port type

Command Usage

Specify out dialing characteristics (tone or pulse) of the channels duration on the port.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} out dialing port type {tone | pulse}**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
	♦ **Syntax Notes** ♦ At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.
slot	Specifies slot number of voice daughtercard installed in switching module, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**)
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).
	♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
tone	Specifies use of Dual Tone Multifrequency (DTMF) dialing.
pulse	Specifies use of rotary pulse dialing.

Default:
The default value is **tone**.

Command Examples:
voice signaling template "sigtempbranch1" out dialing port type tone
voice signaling template "sigtempbranch2" out dialing port type pulse
voice signaling channel 2/1/1-12 out tone interdigit duration 0
voice signaling channel 2/2/13-24 out tone interdigit duration 200
voice signaling channel 2/3/31-30 out tone interdigit duration 2000

Remarks

DTMF consists of eight tones divided into high and low frequency groups for signaling dialed numbers. Each DTMF tone has one high and one low tone each corresponding to a key on a push button dialing pad. Older and less commonly used rotary or circular dials cause breaks in the call circuit flow to signal each number dialed.

voice signaling call duration limit

Command Usage

Specify maximum call time length (channel timing). The call is automatically terminated when the allotted call time length is exceeded.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} [no] call duration limit** <*value*>

<u>Definitions</u>:

TemplateName — Identifies the signaling template by name (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

no — Specifies no call time length (i.e., infinite).

value — Specifies maximum call time length in seconds, from 0 ms to 65,534.

♦ **Syntax Notes** ♦

Do not use commas when entering the maximum call duration limit (for example, **65,534** will return a syntax error message).

A "no call duration limit" means the call can go on forever.

A "no call duration limit 100" or any other number of seconds, means the call can go on forever.

<u>Default</u>:
The default setting is **no call duration limit**.

<u>Command Examples</u>:
voice signaling template "sigtempbranch1" no call duration limit
voice signaling template "sigtempbranch2" call duration limit 0
voice signaling template "sigtempbranch3" no call duration limit 100
voice signaling template "sigtempbranch4" call duration limit 65534
voice signaling channel 2/1/1-12 no call duration limit
voice signaling channel 2/2/13-24 call duration limit 0
voice signaling channel 2/3/1-30 no call duration limit 100
voice signaling channel 2/4/1-30 call duration limit 65534

voice signaling answer wait limit

Command Usage

Specify time to wait for call to be answered (channel timing). The call is automatically terminated if the call destination does not answer within the allotted time period for answering.

Syntax Options

voice signaling { template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} [no] answer wait limit <***value***>**

Definitions:

TemplateName	Identifies the signaling template by name (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ S % ^ _ & | / \ < > ( ) [ ] { }
	♦ **Syntax Notes** ♦ At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).
	♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
no	Specifies no answer wait limit (i.e., infinite).
value	Specifies time to wait before disconnecting a call when there is no answer at the destination, in seconds, from 0 ms to 65,534.
	♦ **Syntax Notes** ♦ *Do not* use commas when entering the no answer wait limit (for example, **65,534** will return a syntax error message). A "no answer wait limit" means wait forever for the call to be answered. A "no answer wait limit 100" or any other number of seconds, means wait forever for calls to be answered.

Default:
The default setting is **no answer wait limit**.

Command Examples:

voice signaling template "sigtempbranch1" no answer wait limit
voice signaling template "sigtempbranch2" answer wait limit 0
voice signaling template "sigtempbranch3" no answer wait limit 100
voice signaling channel 2/1/1-12 no answer wait limit
voice signaling channel 2/2/13-24 answer wait limit 0

voice signaling hang up wait limit

Command Usage

Specify time to wait to force caller to disconnect (channel timing). The call is automatically terminated if the call originator does not hang up within the allotted time period after the destination has hung up.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} [no] hang up wait limit <***value* **>**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

no — Specifies no hang up wait limit (i.e., infinite).

value — Specifies time to wait to before disconnecting a call once the call originator has hung up the phone, in seconds, from 0 ms to 65,534.

♦ **Syntax Notes** ♦

Do not use commas when entering the "no hang up wait limit" (for example, **65,534** will return a syntax error message).

A "no hang up wait limit" means the call remains connected until the call originator hangs up the phone.

A "no hang up wait limit 100" or any other number of seconds, means the call remains connected until the call originator hangs up the phone.

Default:
The default setting is **no hang up wait limit**.

Command Examples:
voice signaling template "sigtempbranch2" hang up wait limit 0
voice signaling channel 2/1/1-12 no hang up wait limit
voice signaling channel 2/2/13-24 hang up wait limit 0

voice signaling fax holdover

Command Usage

Specify time to wait to tear down fax call (channel timing), i.e., set the fax call holdover delay. The delay occurs after an on-hook state is detected during fax mode operation, and before a call clear signal is generated on the line.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** <*slot/port/startChannel-endChannel*} **fax holdover** <*value*>

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies the fax call holdover delay in milliseconds from 0 ms to 65,534, (e.g., **2000**).

♦ **Syntax Note** ♦

Do not use commas when entering the fax call holdover value (for example, **2,000** will return a syntax error message).

Default:

The default value is **2000**.

Command Examples:

voice signaling "sigtempbranch1" fax holdover 2000
voice signaling "sigtempbranch2" fax holdover 0
voice signaling "sigtempbranch3" fax holdover 65534
voice signaling channel 2/1/1-12 fax holdover 2000
voice signaling channel 2/2/13-24 fax holdover 0
voice signaling channel 2/3/1-30 fax holdover 65534

voice signaling companding

Command Usage

Define companding type (Mu Law/A Law) for signaling power at the digital signal processing pulse code modulation (PCM) interface.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **}**
companding {alaw| mulaw}

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { } ♦ **Syntax Notes** ♦ At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**). ♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
alaw	Specifies PCM A Law companding.
mulaw	Specifies PCM Mu Law companding.

Default:
For *VSD only*, the default setting is **mulaw**. For *VSB only*, the default setting is **alaw**.

Command Examples:
voice signaling template "sigtempbranch1" companding mulaw
voice signaling template "sigtempbranch2" companding alaw
voice signaling channel 2/1/1-12 companding mulaw
voice signaling channel 2/2/13-24 companding alaw

voice signaling receive gain

Command Usage

Specify gain in signaling power inserted at *receiver.*

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} receive gain <*gain_value*>**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ Syntax Notes ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ Syntax Note ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

gain_value — Specifies the numerical gain value. Values may range from -14 through 14 (e.g., **-13**, **-2**, **0**, **4**, **13**, etc).

Default:

The default gain value is **0**.

Command Examples:

voice signaling template "sigtempbranch1" receive gain 0
voice signaling channel 2/1/1-12 receive gain 0
voice signaling channel 2/3/1-30 receive gain 14

Remarks

Transmit (TX) and receive (RX) gain is normally used when a device has volume problems. Gain adjusts (increases or decreases) the signal level. Transmit and receive signal gains occur per call on each end of the call. Signal gains are calculated per channel. The signal gains are applied at the channel level. Each channel on a voice switching daughtercard can have a different gain applied. Total gain is calculated between TX/RX points, e.g., if one VSD is set to a gain of -3 and another to a gain of 1, the gain would be -2.

The gain inserted at the receiver comes from the voice switching daughtercard which interprets receive gain as PCM interface-to-packet (H.323) network. This means that the gain is applied to the PCM packet when it is received by the card, and after the packet has been converted to H.323. Analog voice switching daughtercards (VSAs) convert signals to/from PCM before transmit or receive gains are applied.

voice signaling transmit gain

Command Usage

Specify gain in signaling power inserted at *transmitter.*

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} transmit gain <***gain_value***>**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ Syntax Notes ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ Syntax Note ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

gain_value — Specifies the numerical gain value. Values may range from -14 through 14 (e.g., **-13**, **-2**, **0**, **4**, **13**, etc).

Default:
The default gain value is **0**.

Command Examples:
voice signaling "sigtempbranch1" transmit gain 0
voice signaling "sigtempbranch2" transmit gain -14
voice signaling channel 2/2/13-24 transmit gain -14

Remarks

Transmit (TX) and receive (RX) gain is normally used when a device has volume problems. Gain adjusts (increases or decreases) the signal level. Transmit and receive signal gains occur per call on each end of the call. Signal gains are calculated per channel. The signal gains are applied at the channel level. Each channel on a voice switching daughtercard can have a different gain applied. Total gain is calculated between TX/RX points, e.g., if one VSD is set to a gain of -3 and another to a gain of 1, the gain would be -2.

The gain inserted at the transmitter comes from the voice switching daughtercard which interprets transmit gain as (H.323) packet network-to-PCM interface. This means that the gain is applied to the PCM packet when it is transmitted by the card, and after the packet has been converted from H.323. Analog voice switching daughtercards (VSAs) convert signals to/from PCM before transmit or receive gains are applied.

voice signaling idle noise

Command Usage

Specify signaling power amplitude for comfort (idle) noise.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} idle noise <***value***>**

Definitions.

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
	♦ **Syntax Notes** ♦ At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).
	♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
value	Specifies the idle noise level (comfort noise) in 0.01 decibels from -7000 to 7000, (e.g., **1000**); a value of -5000 means -50.
	♦ **Syntax Note** ♦ *Do not* use commas when entering the idle noise value (for example, **1,000** will return a syntax error message).

Default:
The default value is **0**.

Command Examples:
voice signaling template "sigtempbranch1" idle noise 0
voice signaling template "sigtempbranch2" idle noise 1000
voice signaling template "sigtempbranch3" idle noise 7000
voice signaling template "sigtempbranch4" idle noise -7000
voice signaling channel 2/1/1-12 idle noise 0
voice signaling channel 2/2/13-24 idle noise 1000
voice signaling channel 2/3/1-30 idle noise 7000
voice signaling channel 2/4/1-30 idle noise -7000

voice signaling em on hook debounce

Command Usage

Specify E&M signaling time for *transition* (debounce) *to on-hook.*

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} em on hook debounce <***value***>**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ Syntax Notes ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ Syntax Note ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies the E&M debounce (delay interval) transition to on-hook state in milliseconds from 5 to 1,000, (e.g., **50**).

♦ Syntax Note ♦

Do not use commas when entering the E&M transition to on-hook state or debounce transition value (for example, **1,000** will return a syntax error message).

Default:
The default value is **50**.

Command Examples:

voice signaling template "sigtempbranch1" em on hook debounce 50
voice signaling template "sigtempbranch2" em on hook debounce 5
voice signaling template "sigtempbranch3" em on hook debounce 1000
voice signaling channel 2/1/1-12 em on hook debounce 50
voice signaling channel 2/2/13-24 em on hook debounce 5
voice signaling channel 2/3/1-30 em on hook debounce 1000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling) signaling will take effect.

voice signaling em off hook debounce

Command Usage

Specify Ear & Mouth signaling time for *transition* (debounce) *to off-hook.*

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} em off hook debounce <***value***>**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies the E&M debounce (delay interval) transition to off-hook (dial tone) state in milli-seconds from 5 to 1,000, (e.g., **50**).

♦ **Syntax Note** ♦

Do not use commas when entering the E&M off-hook debounce transition value (for example, **1,000** will return a syntax error message).

Default:
The default value is **50**.

Command Examples:
voice signaling template "sigtempbranch1" em off hook debounce 50
voice signaling template "sigtempbranch2" em off hook debounce 5
voice signaling template "sigtempbranch3" em off hook debounce 1000
voice signaling channel 2/1/1-12 em off hook debounce 50
voice signaling channel 2/2/13-24 em off hook debounce 5
voice signaling channel 2/3/1-30 em off hook debounce 1000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling) signaling will take effect.

voice signaling em seize detect

Command Usage

Specify E&M signaling time to wait before *declaring on-hook* (seize detect), i.e., time M-lead must be off-hook before an incoming call is declared.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} em seize detect <***value***>**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ Syntax Notes ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ Syntax Note ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies the *E-lead* seize detect (delay interval) in milliseconds from 5 to 5,000, (e.g., **50**) before an incoming call is declared. *M-lead* must be off-hook before an incoming call can be declared.

♦ Syntax Note ♦

Do not use commas when entering the E&M seize detect value on the E-lead (for example, **5,000** will return a syntax error message).

Default:

The default value is **150**.

Command Examples:

voice signaling template "sigtempbranch1" em seize detect 150
voice signaling template "sigtempbranch3" em seize detect 5000
voice signaling channel 2/1/1-12 em seize detect 150
voice signaling channel 2/2/13-24 em seize detect 5
voice signaling channel 2/3/1-30 em seize detect 5000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling) signaling will take effect.

voice signaling em clear detect

Command Usage

Specify E&M Signaling time to wait before *declaring off-hook* (clear detect), i.e., time off-hook before call clearing is declared.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel***} em clear detect** <*value*>

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , @ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ Syntax Notes ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ Syntax Note ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies the *E-lead* clear detect (delay interval) in milliseconds from 5 to 5,000, (e.g., **50**) before call clearing is declared. *M-lead* needs to be on-hook before call clearing can be declared.

♦ Syntax Note ♦

Do not use commas when entering the E&M clear detect value on the M-lead (for example, **5,000** will return a syntax error message).

Default:
The default value is **400**.

Command Examples:
voice signaling template "sigtempbranch1" em clear detect 400
voice signaling template "sigtempbranch2" em clear detect 5
voice signaling template "sigtempbranch3" em clear detect 5000
voice signaling channel 2/2/13-24 em clear detect 5
voice signaling channel 2/3/1-30 em clear detect 5000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling) signaling will take effect.

voice signaling em clear confirm detect

Command Usage

Specify E&M signaling time to wait before confirming on-hook.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} em clear confirm detect** <*value*>

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ Syntax Notes ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ Syntax Note ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies the *E-lead* clear confirm detect (delay interval) in milliseconds from 5 to 5,000, (e.g., **50**) before call clear confirm is declared.

♦ Syntax Note ♦

Do not use commas when entering the E&M clear confirm detect transition value (for example, **1,000** will return a syntax error message).

Default:

The default value is **5000**.

Command Examples:

voice signaling template "sigtempbranch1" em clear confirm detect 5000
voice signaling template "sigtempbranch2" em clear confirm detect 5
voice signaling template "sigtempbranch3" em clear confirm detect 1000
voice signaling channel 2/1/1-12 em clear confirm detect 5000
voice signaling channel 2/2/13-24 em clear confirm detect 5
voice signaling channel 2/3/ 1-30 em clear confirm detect 1000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling) signaling will take effect.

voice signaling em clear confirm wait max

Command Usage

Specify Ear & Mouth signaling time to wait for on-hook after a clear.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel***} em clear confirm wait max[imum] <***value***>**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & I / \ < > ( ) [ ] { }
	♦ **Syntax Notes** ♦ At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).
	♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
imum	Optional command syntax. You can type either **max** or **maximum** in the command line.
value	Specifies the maximum duration (delay interval) to wait for an on-hook response on the *M-lead* after going on-hook on the *E-lead*, (e.g., **25000**).
	♦ **Syntax Note** ♦ *Do not* use commas when entering the E&M on-hook after a clear detect value (for example, **60,000** will return a syntax error message).

Default:
The default value is **60000**.

Command Examples:

voice signaling template "sigtempbranch1" em clear confirm wait maximum 60000
voice signaling template "sigtempbranch2" em clear confirm wait max 25000
voice signaling template "sigtempbranch3" em clear confirm wait max 45000
voice signaling channel 2/1/1-12 em clear confirm wait maximum 60000
voice signaling channel 2/2/13-24 em clear confirm wait max 25000
voice signaling channel 2/3/1-30 em clear confirm wait max 45000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling) signaling will take effect.

voice signaling em guard all

Command Usage

Specify Ear & Mouth signaling time to wait between termination and *origination.*

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} em guard all** <*value*>

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — After an aborted call, specifies the time period (delay interval) in milliseconds, (e.g., **400**) from 0 to 60,000, when neither incoming nor outgoing calls are accepted or initiated.

♦ **Syntax Note** ♦

Do not use commas when entering the E&M time span (in which no incoming or outgoing calls are accepted or initiated) for call termination and origination value (for example, **10,000** will return a syntax error message).

Default:

The default value is **400**.

Command Examples:

voice signaling template "sigtempbranch1" em guard all 400
voice signaling template "sigtempbranch2" em guard all 20000
voice signaling template "sigtempbranch3" em guard all 60000
voice signaling channel 2/1/1-12 em guard all 400
voice signaling channel 2/2/13-24 em guard all 20000
voice signaling channel 2/3/1-30 em guard all 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling) signaling will take effect.

voice signaling em guard out

Command Usage

Specify Ear & Mouth signaling time to wait between termination and *receiving*.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} em guard out <***value***>**

Definitions.

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies the time period (delay interval) in milliseconds, (e.g., **400**) from 0 to 60,000, when only incoming calls are accepted or initiated. Outgoing calls are aborted.

♦ **Syntax Note** ♦

Do not use commas when entering the E&M extended time span (in which only incoming calls are accepted or initiated) for call termination and receiving value (for example, **10,000** will return a syntax error message).

Default:

The default value is **400**.

Command Examples:

voice signaling template "sigtempbranch1" em guard out 400
voice signaling template "sigtempbranch2" em guard out 20000
voice signaling template "sigtempbranch3" em guard out 60000
voice signaling channel 2/1/1-12 em guard out 400
voice signaling channel 2/2/13-24 em guard out 20000
voice signaling channel 2/3/1-30 em guard out 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling) signaling will take effect.

voice signaling em dial tone

Command Usage

Set Ear & Mouth signaling dial tone generation on incoming calls (on/off).

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} em dial tone {on | off}**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
	♦ **Syntax Notes** ♦ At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).
	♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
on	Turns ON E&M dial tone generation for incoming calls.
off	Turns OFF E&M dial tone generation for incoming calls.

Default:
The default setting is **none**.

Command Examples:
voice signaling template "sigtempbranch1" em dial tone on
voice signaling template "sigtempbranch2" em dial tone off
voice signaling channel 2/1/1-12 em dial tone on
voice signaling channel 2/2/13-24 em dial tone off
voice signaling channel 2/3/1-30 em dial tone on

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling) signaling will take effect.

voice signaling em min connection time

Command Usage

Specify minimum Ear & Mouth signaling connection time.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} em min[imum] connect time <***value***>**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { } ♦ **Syntax Notes** ♦ At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**). ♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
imum	Optional command syntax. You can type either **min** or **minimum** in the command line.
value	Specifies the minimum time period, in milliseconds from 0 to 20,000, (e.g., **2000**), that a connection is maintained. ♦ **Syntax Note** ♦ *Do not* use commas when entering the minimum E&M connection value (for example, **2,000** will return a syntax error message).

Default:
The default value is **2000**.

Command Examples:
voice signaling template "sigtempbranch1" em minimum connection time 50
voice signaling channel 2/3/1-30 em minimum connection time 1000

Remarks

If the remote (or called) end disconnects during the specified minimum E&M connection time, the disconnect is acknowledged by a dial tone.

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling) signaling will take effect.

voice signaling em hang up wait

Command Usage

Specify time to wait after Ear & Mouth signaling hang up before disabling.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} em hang up wait <***value***>**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
	♦ **Syntax Notes** ♦ At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).
	♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
value	Specifies the time period, in milliseconds, from 5 to 60,000, (e.g., **2000**) a port waits to hang up before signaling that the call is in a disabled state.
	♦ **Syntax Note** ♦ *Do not* use commas when entering the value for E&M time to wait after signaling call hang up before disabling (for example, **2,000** will return a syntax error message).

Default:
The default value is **2000**.

Command Examples:
voice signaling template "sigtempbranch1" em hang up wait 2000
voice signaling template "sigtempbranch2" em hang up wait 10000
voice signaling template "sigtempbranch3" em hang up wait 20000
voice signaling channel 2/1/1-12 hang up wait 2000
voice signaling channel 2/2/13-24 em hang up wait 10000
voice signaling channel 2/2/1-30 em hang up wait 20000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling) signaling will take effect.

voice signaling emw in wink wait min

Command Usage

Specify *minimum* Ear & Mouth wink delay on incoming calls.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} emw in wink wait min[imum] <***value* **>**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
	♦ Syntax Notes ♦ At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).
	♦ Syntax Note ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
imum	Optional command syntax. You can type either **min** or **minimum** in the command line.
value	Specifies the minimum delay in milliseconds, from 5 to 60,000, (e.g., **150**), before beginning the wink on the *E-lead* after detecting a line seizure on the *M-lead* for incoming calls.
	♦ Syntax Note ♦ *Do not* use commas when entering the minimum E&M wink delay value on incoming calls (for example, **24,000** will return a syntax error message).

Default:
The default value is **2000**.

Command Examples:

voice signaling template "sigtempbranch1" emw in wink wait minimum 150
voice signaling template "sigtempbranch2" emw in wink wait min 24000
voice signaling template "sigtempbranch3" emw in wink wait min 60000
voice signaling channel 2/1/1-12 emw in wink wait minimum 150
voice signaling channel 2/2/13-24 emw in wink wait min 30000
voice signaling channel 2/3/1-30 emw in wink wait min 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling emw in wink wait max

Command Usage

Specify *maximum* Ear & Mouth wink delay on incoming calls.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} emw in wink wait max[imum]** <*value*>

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

imum — Optional command syntax. You can type either **max** or **maximum** in the command line.

value — Specifies the maximum delay in milliseconds, from 5 to 60,000, (e.g., **3000**), before beginning the wink on the *E-lead* after detecting a line seizure on the *M-lead* for incoming calls.

♦ **Syntax Note** ♦

Do not use commas when entering the maximum E&M wink delay value on incoming calls (for example, **3,000** will return a syntax error message).

Default:

The default value is **3000**.

Command Examples:

voice signaling template "sigtempbranch1" emw in wink wait maximum 3000
voice signaling template "sigtempbranch2" emw in wink wait max 30000
voice signaling template "sigtempbranch3" emw in wink wait max 60000
voice signaling channel 2/1/1-12 emw in wink wait maximum 3000
voice signaling channel 2/2/13-24 emw in wink wait max 30000
voice signaling channel 2/3/1-30 emw in wink wait max 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling emw in wink duration

Command Usage

Specify duration of Ear & Mouth wink delay on incoming calls.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} emw in wink duration** <*value*>

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ Syntax Notes ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ Syntax Note ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies the duration of the wink signal on the *E-lead* for incoming calls in milliseconds from 5 to 60,000, (e.g., **200**).

♦ Syntax Note ♦

Do not use commas when entering the E&M wink duration value for incoming calls (for example, **200** will return a syntax error message).

Default:
The default value is **200**.

Command Examples:

voice signaling template "sigtempbranch1" emw in wink duration 200
voice signaling template "sigtempbranch2" emw in wink duration 30000
voice signaling template "sigtempbranch3" emw in wink duration 60000
voice signaling channel 2/1 1-12 emw in wink duration 200
voice signaling channel 2/2 13-24 emw in wink duration 30000
voice signaling channel 2/3 1-30 emw in wink duration 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling emw in wink digit ignore

Command Usage

Specify time to ignore tone (digits) after Ear & Mouth wink.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} emw in wink digit ignore <***value***>**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies the period of time, in milliseconds from 5 to 1,000, (e.g., **30**)., after the wink to ignore tones (digits) for incoming calls.

♦ **Syntax Note** ♦

Do not use commas when entering the value for E&M time to ignore digits for incoming after wink (for example, **1,000** will return a syntax error message).

Default:

The default value is **30** milliseconds.

Command Examples:

voice signaling template "sigtempbranch1" emw in wink digit ignore 30
voice signaling template "sigtempbranch2" emw in wink digit ignore 5
voice signaling template "sigtempbranch3" emw in wink digit ignore 1000
voice signaling channel 2/1 1-12 emw in wink digit ignore 30
voice signaling channel 2/2 13-24 emw in wink digit ignore 5
voice signaling channel 2/3 1-30 emw in wink digit ignore 1000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling emw out winkwait max

Command Usage

Specify time to wait for E&M wink on outgoing calls.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel***} emw out winkwait max[imum] <***value***>**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

imum — Optional command syntax. You can type either **max** or **maximum** in the command line.

value — Specifies the maximum time to wait for a wink response on the *M-lead* after going off-hook on the *E-lead*, in milliseconds from 5 to 60,000, (e.g., **8000**).

♦ **Syntax Note** ♦

Do not use commas when entering the E&M maximum time to wait value for a wink response on outgoing calls (for example, **8,000** will return a syntax error message).

Default:

The default value is **8000** milliseconds.

Command Examples:

voice signaling template "sigtempbranch1" emw out winkwait maximum 8000
voice signaling template "sigtempbranch2" emw out winkwait max 30000
voice signaling template "sigtempbranch4" emw out winkwait max 60000
voice signaling channel 2/1/1-12 emw out winkwait maximum 8000
voice signaling channel 2/1/13-24 emw out winkwait max 30000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling emw out wink duration min

Command Usage

Specify *minimum* E&M wink duration.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} emw out wink duration min[imum]** <*value*>

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { } ♦ **Syntax Notes** ♦ At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**). ♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
imum	Optional command syntax. You can type either **min** or **minimum** in the command line.
value	Specifies the minimum duration of the wink response to the *M-lead* for detection by the *M-lead*, in milliseconds from 5 to 60,000, (e.g., **100**). ♦ **Syntax Note** ♦ *Do not* use commas when entering the value for the minimum duration of the wink response to the M-lead for detection (for example, **1,000** will return a syntax error message).

Default:
The default value is **100** milliseconds.

Command Examples:

voice signaling template "sigtempbranch1" emw out wink duration minimum 100
voice signaling template "sigtempbranch2" emw out wink duration min 30000
voice signaling template "sigtempbranch3" emw out wink duration min 60000
voice signaling channel 2/1/1-12 emw out wink duration min 100
voice signaling channel 2/2/13-24 emw out wink duration min 30000
voice signaling channel 2/3/1-30 emw out wink duration min 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol-type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling emw out wink duration max

Command Usage

Specify *maximum* E&M wink duration.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel***} emw out wink duration max[imum]** <*value*>

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ` ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

imum — Optional command syntax. You can type either **max** or **maximum** in the command line.

value — Specifies the maximum duration of the wink response on the *M-lead* before glare condition (trunk ends are seized simultaneously) declared on outgoing call, in milliseconds from 5 to 60,000, (e.g., **800**).

♦ **Syntax Note** ♦

Do not use commas when entering the value for the maximum duration of the wink response to the M-lead for detection (for example, **1,000** will return a syntax error message).

Default:
The default value is **800** milliseconds.

Command Example:
voice signali

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

Use this command to reduce instances of trunk deadlock caused by trunks which are seized on both ends of a call at the same time.

voice signaling emi glare report

Command Usage

Specify E&M immediate start time to remain off-hook when congested.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} emi glare report <***value***>**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies time period, in milliseconds from 0 to 60,000, (e.g., **5000**), wherein if glare condition is detected, the line stays off-hook and generates a congestion tone.

♦ **Syntax Note** ♦

Do not use commas when entering the E&M immediate start-time value in which a glare condition is reported and a congestion tone is generated while off-hook (for example, **5,000** will return a syntax error message).

Default:

The default value is **5000** milliseconds.

Command Examples:

voice signaling template "sigtempbranch1" emi glare report 5000
voice signaling template "sigtempbranch2" emi glare report 30000
voice signaling template "sigtempbranch3" emi glare report 60000
voice signaling channel 2/1/1-12 emi glare report 5000
voice signaling channel 2/2/13-24 emi glare report 30000
voice signaling channel 2/3/1-30 emi glare report 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling); and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling emi digit wait

Command Usage

Specify E&M immediate start time to wait before beginning digit collection.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} emi digit wait <***value***>**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦
At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦
Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies time period, in milliseconds from 0 to 60,000, (e.g., **200**), to wait for the voice daughtercard to be ready before collecting digits. The line stays off-hook and generates a dial tone if glare condition detected in the interim.

♦ **Syntax Note** ♦
Do not use commas when entering the E&M immediate start time value, i.e., time to wait for before digit collection is enabled and an off-hook congestion tone with glare condition is reported (for example, **5,000** will return a syntax error message).

Default:
The default value is **200** milliseconds.

Command Examples:
voice signaling template "sigtempbranch1" emi digit wait 200
voice signaling template "sigtempbranch2" emi digit wait 30000
voice signaling channel 2/1/1-12 emi digit wait 200
voice signaling channel 2/2/13-24 emi digit wait 30000
voice signaling channel 2/3/1-30 emi digit wait 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling emd in delay min

Command Usage

Specify minimum E&M delay start response to off-hook (dial tone) state.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} emd in delay min[imum]** <*value*>

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ Syntax Notes ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ Syntax Note ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

imum — Optional command syntax. You can type either **min** or **minimum** in the command line.

value — Specifies the minimum duration of the delay signal response to the seize detect of incoming calls on the *M-lead*, in milliseconds from 0 to 60,000, (e.g., **200**).

♦ Syntax Note ♦

Do not use commas when entering the minimum value for E&M delay start response to off-hook (dial tone) state (for example, **5,000** will return a syntax error message).

Default:

The default value is **200** milliseconds.

Command Examples:

voice signaling template "sigtempbranch1" emd in delay minimum 200
voice signaling template "sigtempbranch2" emd in delay min 30000
voice signaling template "sigtempbranch3" emd in delay min 60000
voice signaling channel 2/1/-15 emd in delay minimum 200
voice signaling channel 2/2/13-24 emd in delay min 30000
voice signaling channel 2/3/1-30 emd in delay min 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling emd in delay max

Command Usage

Specify maximum E&M delay start response to off-hook (dial tone) state.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} emd in delay max[imum]** <*value*>

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
	♦ **Syntax Notes** ♦ At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).
	♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
imum	Optional command syntax. You can type either **max** or **maximum** in the command line.
value	Specifies the maximum duration of the delay signal response to the seize detect of incoming calls on the *M-lead*, in milliseconds from 5 to 60,000, (e.g., **2500**).
	♦ **Syntax Note** ♦ *Do not* use commas when entering the maximum value for E&M delay start response to off-hook (dial tone) state (for example, **5,000** will return a syntax error message).

Default:
The default value is **2500** milliseconds.

Command Examples:

voice signaling template "sigtempbranch1" emw in delay maximum 2500
voice signaling template "sigtempbranch2" emw in delay max 30000
voice signaling template "sigtempbranch3" emw in delay max 60000
voice signaling channel 2/1/1-12 emw in delay maximum 2500
voice signaling channel 2/2/13-24 emw in delay max 30000
voice signaling channel 2/3/1-30 emw in delay max 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling emd in digit ignore

Command Usage

Specify time to ignore incoming digits after E&M delay start.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} emd in digit ignore** <*value*>

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies the period of time after which E&M delay signal is completed before digits on incoming calls are accepted, in milliseconds from 5 to 60,000, (e.g., **2500**).

♦ **Syntax Note** ♦

Do not use commas when entering the value for time to ignore incoming digits after E&M delay start (for example, **5,000** will return a syntax error message).

Default:

The default value is **2500** milliseconds.

Command Examples:

voice signaling template "sigtempbranch1" emd in digit ignore 2500
voice signaling template "sigtempbranch2" emd in digit ignore 30000
voice signaling template "sigtempbranch3" emd in digit ignore 60000
voice signaling template channel 2/1/1-12 emd in digit ignore 2500
voice signaling template channel 2/2/13-24 emd in digit ignore 30000
voice signaling template channel 2/3/1-30 emd in digit ignore 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling emd out integrity check

Command Usage

Specify E&M delay start signal detection.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} emd out integrity check {on | off}**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ Syntax Notes ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ Syntax Note ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

on — Turns ON E&M delay integrity check on outgoing calls.

off — Turns OFF E&M delay integrity check on outgoing calls.

Default:

The default setting is **off**.

Command Examples:

voice signaling template "sigtempbranch1" emd out integrity check off
voice signaling template "sigtempbranch2" emd out integrity check on
voice signaling channel 2/1/1-12 emd out integrity check off
voice signaling channel 2/2/13-24 emd out integrity check on

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

When the Integrity Check mode is ON, the delay signal response is required from the PBX for outgoing calls.

voice signaling emd out delay duration min

Command Usage

Specify *minimum* E&M delay start detection time on "M" lead.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} emd out delay duration min[imum]** <*value*>

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
	♦ **Syntax Notes** ♦ At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).
	♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
imum	Optional command syntax. You can type either **min** or **minimum** in the command line.
value	Specifies the minimum duration of the delay signal response on the *M-lead* for detection on outgoing calls, in milliseconds from 5 to 60,000, (e.g., **100**).
	♦ **Syntax Note** ♦ *Do not* use commas when entering the value for the minimum E&M delay start detection on the M-lead signal (for example, **1,000** will return a syntax error message).

Default:
The default value is **100** milliseconds.

Command Examples:

voice signaling template "sigtempbranch1" emd out delay duration minimum 100
voice signaling template "sigtempbranch2" emd out delay duration min 30000
voice signaling template "sigtempbranch3" emd out delay duration min 60000
voice signaling channel 2/1/1-12 emd out delay duration minimum 100
voice signaling channel 2/2/13-24 emd out delay duration min 30000
voice signaling channel 2/3/1-30 emd out delay duration min 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling emd out detail duration max

Command Usage

Specify *maximum* E&M delay start detection time on "M" lead.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} emd out delay duration max[imum] <***value***>**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

imum — Optional command syntax. You can type either **max** or **maximum** in the command line.

value — Specifies the maximum duration of the delay signal response on the *M-lead* before a glare condition on outgoing calls can be declared, in milliseconds from 5 to 60,000, (e.g., **100**).

♦ **Syntax Note** ♦

Do not use commas when entering the value for the maximum E&M delay signal response on the M-lead before a glare condition can be declared (for example, **1,000** will return a syntax error message).

Default:

The default value is **8000** milliseconds.

Command Examples:

voice signaling template "sigtempbranch1" emd out delay duration maximum 8000
voice signaling template "sigtempbranch2" emd out delay duration max 30000
voice signaling channel 2/1/1-12 emd out delay duration maximum 8000
voice signaling channel 2/2/13-24 emd out delay duration max 3000
voice signaling channel 2/3/1-30 emd out delay duration max 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling emd out delay check

Command Usage

Specify *maximum* time to wait for E&M delay start detection.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel***} emd out delay check <***value***>**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ Syntax Notes ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ Syntax Note ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies the period of time after going off-hook on the *E-lead* before checking the *M-lead* for the delay signal response, in milliseconds from 5 to 60,000, (e.g., **170**). If the response is not detected in the interim, the call setup process resumes immediately.

♦ Syntax Note ♦

Do not use commas when entering the value for the maximum time to wait for E&M delay start detection (for example, **5,000** will return a syntax error message).

Default:
The default value is **170** milliseconds.

Command Examples:
voice signaling template "sigtempbranch1" emd out delay check 170
voice signaling channel 2/1/1-12 emd out delay check 170
voice signaling channel 2/2/13-24 emd out delay check 30000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

In order to use this command, the integrity check for outgoing calls must be turned OFF via the **voice signaling emd out integrity check** command.

voice signaling fxs ls on hook debounce

Command Usage

Specify Foreign Exchange Station Loop Start (FXS LS) debounce interval to *on-hook* transition.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel***} fxs ls on hook debounce <***value***>**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ` ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies the debounce (delay interval) transition to on-hook state in milliseconds from 1 to 1,000, (e.g., **20**).

♦ **Syntax Note** ♦

Do not use commas when entering the fxs ls on-hook debounce transition value (for example, **1,000** will return a syntax error message).

Default:

The default value is **20** milliseconds.

Command Examples:

voice signaling template "sigtempbranch1" fxs ls on hook debounce 20
voice signaling template "sigtempbranch2" fxs ls on hook debounce 500
voice signaling template "sigtempbranch3" fxs ls on hook debounce 1000
voice signaling channel 2/1/1-12 fxs ls on hook debounce 200
voice signaling channel 2/2/13-24 fxs ls on hook debounce 500
voice signaling channel 2/3/1-30 fxs ls on hook debounce 1000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs ls off hook debounce

Command Usage

Specify Foreign Exchange Station Loop Start (FXS LS) debounce interval to *off-hook* transition.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxs ls off hook debounce <***value***>**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies the debounce (delay interval) transition to off-hook (dial tone) state in milliseconds from 1 to 1,000, (e.g., **20**).

♦ **Syntax Note** ♦

Do not use commas when entering the fxs ls off-hook debounce transition value (for example, **1,000** will return a syntax error message).

Default:
The default value is **20** milliseconds.

Command Examples:

voice signaling template "sigtempbranch1" fxs ls off hook debounce 20
voice signaling template "sigtempbranch2" fxs ls off hook debounce 500
voice signaling template "sigtempbranch3" fxs ls off hook debounce 1000
voice signaling channel 2/1/1-12 fxs ls off hook debounce 20
voice signaling channel 2/2/13-24 fxs ls off hook debounce 500
voice signaling channel 2/3/1-30 fxs ls off hook debounce 1000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs ls seize detect

Command Usage

Specify Foreign Exchange Station Loop Start (FXS LS) time to wait before declaring off-hook.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxs ls seize detect <***value***>**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _& | / \ < > ( ) [ ] { }
	♦ **Syntax Notes** ♦ At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).
	♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
value	Specifies the time to wait before off-hook condition is declared, in milliseconds from 1 to 5,000, (e.g., **150**).
	♦ **Syntax Note** ♦ *Do not* use commas when entering the fxs ls time to wait value before declaring off-hook condition (for example, **1,000** will return a syntax error message).

Default:

The default value is **150** milliseconds.

Command Examples:

voice signaling template "sigtempbranch1" fxs ls seize detect 150
voice signaling template "sigtempbranch2" fxs ls seize detect 2500
voice signaling template "sigtempbranch3" fxs ls seize detect 5000
voice signaling channel 2/1/1-12 fxs ls seize detect 150
voice signaling channel 2/2/13-24 fxs ls seize detect 2500
voice signaling channel 2/3/1-30 fxs ls seize detect 5000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs ls originate clear detect

Command Usage

Specify Foreign Exchange Station Loop Start (FXS LS) minimum time to wait before declaring on-hook by *originator.*

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel***} fxs ls originate clear detect <***value***>**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ Syntax Notes ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ Syntax Note ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies the minimum time to wait if call originator hangs up before on-hook condition declared, in milliseconds from 1 to 60,000, (e.g., **300**).

♦ Syntax Note ♦

Do not use commas when entering the fxs ls call originate clear detect value (for example, **1,000** will return a syntax error message).

Default:
The default value is **300** milliseconds.

Command Examples:
voice signaling template "sigtempbranch1" fxs ls originate clear detect 300
voice signaling template "sigtempbranch2" fxs ls originate clear detect 30000
voice signaling template "sigtempbranch3" fxs ls originate clear detect 60000
voice signaling channel 2/1/1-12 fxs ls originate clear detect 300
voice signaling channel 2/1/13-24 fxs ls originate clear detect 30000
voice signaling channel 2/1/1-30 fxs ls originate clear detect 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs ls answer clear detect

Command Usage

Specify Foreign Exchange Station Loop Start (FXS LS) minimum time to wait before declaring on-hook by *answerer.*

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxs ls answer clear detect <***value* **>**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value	Specifies the minimum time to wait if answering party hangs up before on-hook condition declared, in milliseconds from 1 to 60,000, (e.g., **300**).

♦ **Syntax Note** ♦

Do not use commas when entering the fxs ls answering party clear detect value (for example, **1,000** will return a syntax error message).

Default:
The default value is **300** milliseconds.

Command Examples:
voice signaling template "sigtempbranch1" fxs ls answer clear detect 300
voice signaling template "sigtempbranch2" fxs ls answer clear detect 30000
voice signaling template "sigtempbranch3" fxs ls answer clear detect 60000
voice signaling channel 2/1/1-12 fxs ls answer clear detect 300
voice signaling channel 2/2/13-24 fxs ls answer clear detect 30000
voice signaling channel 2/3/1-30 fxs ls answer clear detect 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs ls supervisory disconnect wait

Command Usage

Specify Foreign Exchange Station Loop Start (FXS LS) time to wait after supervisory disconnect before declaring on-hook.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel***} fxs ls supervisory disconnect wait <***value* **>**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { } ♦ **Syntax Notes** ♦ At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**). ♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
value	Specifies the fxs ls supervisory disconnect (CPC signal) wait value, in milliseconds from 1 to 60,000, (e.g., **200**) is generated before on-hook condition declared. ♦ **Syntax Note** ♦ *Do not* use commas when entering the fxs ls supervisory disconnect wait value (for example, **1,000** will return a syntax error message).

Default:
The default value is **200** milliseconds.

Command Examples:
voice signaling template "sigtempbranch1" fxs ls supervisory disconnect wait 200
voice signaling template "sigtempbranch2" fxs ls supervisory disconnect wait 30000
voice signaling template "sigtempbranch3" fxs ls supervisory disconnect wait 60000
voice signaling channel 2/1/1-12 fxs ls supervisory disconnect wait 200
voice signaling channel 2/2/16-24 fxs ls supervisory disconnect wait 30000
voice signaling channel 2/3/1-30 fxs ls supervisory disconnect wait 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs ls supervisory disconnect duration

Command Usage

Specify Foreign Exchange Station Loop Start (FXS LS) duration of supervisory disconnect.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxs ls supervisory disconnect duration <***value* **>**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ Syntax Notes ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ Syntax Note ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies the fxs ls supervisory disconnect (CPC signal) wait value, in milliseconds from 1 to 60,000, (e.g., **850**) generated before on-hook condition declared.

♦ Syntax Note ♦

Do not use commas when entering the fxs ls supervisory disconnect wait value (for example, **10,000** will return a syntax error message).

Default:

The default value is **850** milliseconds.

Command Examples:

voice signaling template "sigtempbranch1" fxs ls supervisory disconnect duration 850
voice signaling template "sigtempbranch2" fxs ls supervisory disconnect duration 30000
voice signaling template "sigtempbranch3" fxs ls supervisory disconnect duration 60000
voice signaling channel 2/1/1-12 fxs ls supervisory disconnect duration 850
voice signaling channel 2/2/13-24 fxs ls supervisory disconnect duration 30000
voice signaling channel 2/3/1-30 fxs ls supervisory disconnect duration 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs ls caller id

Command Usage

Set Foreign Exchange Station Loop Start (FXS LS) to *generate outbound* caller ID (on/off).

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxs ls caller id {on | off}**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦
At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦
Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

on — Turns ON fxs ls caller ID for specified signaling template or port.

off — Turns OFF fxs ls caller ID for specified signaling template or port.

♦ **Syntax Note** ♦
The **voice coding profile caller id** command must be enabled to use this command.

Default:
The default setting is **off**.

Command Examples:
voice signaling template "sigtempbranch1" fxs ls caller id off
voice signaling template "sigtempbranch2" fxs ls caller id on
voice signaling template channel 2/1/1-12 fxs ls caller id off
voice signaling template channel 2/2/13-24 fxs ls caller id on

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs ls cadence coefficient

Command Usage

Set Foreign Exchange Station Loop Start (FXS LS) cadence coefficient, or continental ring tone (North America/Europe).

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxs ls cadence coefficient {north america | europe}**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & ! / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦
At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦
Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

north america — Turns ON fxs ls cadence coefficient for North America.

europe — Turns ON fxs ls cadence coefficient for Europe.

♦ **Syntax Note** ♦
This command must be set in relation to the **voice signaling fxs ls ring ID** command.

Default:
The default setting is **north america**.

Command Examples:
voice signaling template "sigtempbranch1" fxs ls cadence coefficient north america
voice signaling template "sigtempbranch2" fxs ls cadence coefficient europe
voice signaling channel 2/1/1-12 fxs ls cadence coefficient north america
voice signaling channel 2/2/13-24 fxs ls cadence coefficient europe

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs ls ring id

Command Usage

Set Foreign Exchange Station Loop Start (FXS LS) ring ID, or continental ring tone variance for North America or Europe.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxs ls ring id {0 | 1 | 2 | 3 | 4 | 5 | 6 | 7 | default}**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

0, 1, 2, 3, 4, 5, 6, 7 — Specifies variance in continental ring tone for either North America or Europe (e.g., **2**) as per selected coefficient. The value **0** is the same as **default**.

default — Sets the tone variance to the default setting, i.e, 0.

♦ **Syntax Note** ♦

This command must be set in relation to the **voice signaling fxs ls cadence coefficient** command.

Default:

The default setting is **0**.

Command Examples:

voice signaling template "sigtempbranch1" fxs ls ring ID default
voice signaling template "sigtempbranch2" fxs ls ring ID 0
voice signaling template "sigtempbranch3" fxs ls ring ID 7
voice signaling channel 2/1/1-12 fxs ls ring ID default
voice signaling channel 2/2/13-24 fxs ls ring ID 0
voice signaling channel 2/3/1-30 fxs ls ring ID 7

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

The ring cadence that is set for the first channel on the first port is automatically set for all remaining FXS/FXO ports.

All ports must have the same ring cadence.

On the Omni Access 512, the ring cadence can only be specified for port 1.

On the Omni Switch/Router, the ring cadence can only be specified for port 1 (or 9 for a dual VSA-FXS/FXS in an HSX-H).

voice signaling fxo ls ringing debounce

Command Usage

Specify Foreign Exchange Office Loop Start (FXO LS) incoming ring signal debounce interval.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxo ls ringing debounce <***value***>**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies the fxo ls incoming ring signal debounce (delay interval) in milliseconds from 1 to 1,000, (e.g., **50**).

♦ **Syntax Note** ♦

Do not use commas when entering the fxo ls incoming ring signal debounce value (for example, **1,000** will return a syntax error message).

Default:

The default value is **50** milliseconds.

Command Examples:

voice signaling template "sigtempbranch1" fxo ls ringing debounce 50
voice signaling template "sigtempbranch2" fxo ls ringing debounce 500
voice signaling template "sigtempbranch3" fxo ls ringing debounce 1000
voice signaling channel 2/1/1-12 fxo ls ringing debounce 50
voice signaling channel 2/2/13-24 fxo ls ringing debounce 500
voice signaling channel 2/3/1-30 fxo ls ringing debounce 1000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo ls supervisory disconnect detection

Command Usage

Set Foreign Exchange Office Loop Start (FXO LS) *supervisory disconnect* detection signal (enable/disable).

Syntax Options

voice signaling {template *"TemplateName"* | channel *slot/port/startChannel-endChannel*} fxo ls supervisory disconnect detection {on | off}

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & ! / \ < > ( ) [ ] { } ♦ **Syntax Notes** ♦ At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**). ♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
on	Turns ON fxo ls detection of supervisory disconnect for specified signaling template or port.
off	Turns OFF fxo ls detection of supervisory disconnect for specified signaling template or port. ♦ **Syntax Note** ♦ To use this command properly, the duration of the fxo ls supervisory disconnect signal must be specified via the **voice signaling fxo ls supervisory disconnect** command.

Default:
The default setting is **on**.

Command Examples:
voice signaling template "sigtempbranch1" fxo ls supervisory disconnect detection on
voice signaling channel 2/1/1-12 fxo ls supervisory disconnect detection on
voice signaling channel 2/2/13-24 fxo ls supervisory disconnect detection off

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo ls supervisory disconnect

Command Usage

Specify Foreign Exchange Office (FXO) Loop Start *duration of supervisory disconnect* detection signal.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxo ls supervisory disconnect <***value***>**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies the fxo ls incoming ring signal debounce (delay interval) in milliseconds from **1** to 60,000, (e.g., **600**). If the loop current drops below the specified value for a period of time, it is not considered a "supervisory disconnect" of the signal.

♦ **Syntax Notes** ♦

Do not use commas when entering the duration of the fxo ls supervisory disconnect (CPC signal) value (for example, **10,000** will return a syntax error message).

To use this command, detection of the fxo ls supervisory disconnect signal must be turned ON via the **voice signaling fxo ls supervisory disconnect detection** command. If the fxo ls supervisory disconnect signal detection is turned OFF, the disconnect signal is set automatically to a value of 65535.

Default:
The default value is **600** milliseconds.

Command Examples:
voice signaling template "sigtempbranch1" fxo ls supervisory disconnect 600
voice signaling template "sigtempbranch2" fxo ls supervisory disconnect 30000
voice signaling template "sigtempbranch3" fxo ls supervisory disconnect 60000
voice signaling channel 2/3/1-30 fxo ls supervisory disconnect 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo ls guard out

Command Usage

Specify Foreign Exchange Office (FXO) Loop Start before originating calls while receiving calls.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxo ls guard out <***value***>**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { } ♦ **Syntax Notes** ♦ At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**). ♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
value	Specifies the fxo ls incoming ring signal debounce (delay interval) in milliseconds from 1 to 1,000, (e.g., **50**). ♦ **Syntax Notes** ♦ *Do not* use commas when entering the duration of the fxo ls supervisory disconnect (CPC signal) value (for example, **10,000** will return a syntax error message). To use this command, detection of the fxo ls supervisory disconnect signal must be turned ON via the **voice signaling fxo ls supervisory disconnect detection** command.

Default:
The default value is **50** milliseconds.

Command Examples:
voice signaling template "sigtempbranch1" fxo ls guard out 50
voice signaling template "sigtempbranch2" fxo ls guard out 5
voice signaling template "sigtempbranch3" fxo ls guard out 100
voice signaling channel 2/1/1-12 fxo ls guard out 50
voice signaling channel 2/2/13-24 fxo ls guard out 5
voice signaling channel 2/2/1-30 fxo ls guard out 1000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/ FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo ls ringing inter cycle

Command Usage

Specify Foreign Exchange Office (FXO) Loop Start time between ring *cycles* to detect ringing.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxo ls ringing inter cycle <***value***>**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ Syntax Notes ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ Syntax Note ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies the time period between ring cycles (ringing pulse plus time between ring pulses) to detect ringing, in milliseconds from 1 to 60,000, (e.g., **2000**)

♦ Syntax Note ♦

Do not use commas when entering the fxo ls between ring cycles ring detection value (for example, **10,000** will return a syntax error message).

Default:

The default value is **2000** milliseconds.

Command Examples:

voice signaling template "sigtempbranch1" fxo ls ringing inter cycle 2000
voice signaling template "sigtempbranch2" fxo ls ringing inter cycle 30000
voice signaling template "sigtempbranch3" fxo ls ringing inter cycle 60000
voice signaling channel 2/1/1-12 fxo ls ringing inter cycle 2000
voice signaling channel 2/2/13-24 fxo ls ringing inter cycle 30000
voice signaling channel 2/3/1-30 fxo ls ringing inter cycle 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/ FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo ls ringing inter pulse

Command Usage

Specify Foreign Exchange Office (FXO) Loop Start time between ring *pulses* to detect ringing.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxo ls ringing inter pulse** <*value*>

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ Syntax Notes ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ Syntax Note ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies the time period between ring pulses to detect ringing, in milliseconds from 1 to 60,000, (e.g., **550**).

♦ Syntax Note ♦

Do not use commas when entering the fxo ls between ring pulses ring detection value (for example, **10,000** will return a syntax error message).

Default:

The default value is **550** milliseconds.

Command Examples:

voice signaling template "sigtempbranch1" fxo ls ringing inter pulse 550
voice signaling template "sigtempbranch2" fxo ls ringing inter pulse 30000
voice signaling template "sigtempbranch3" fxo ls ringing inter pulse 60000
voice signaling channel 2/1/1-12 fxo ls ringing inter pulse 550
voice signaling channel 2/2/13-24 fxo ls ringing inter pulse 30000
voice signaling channel 2/3/1-30 fxo ls ringing inter pulse 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo ls caller id

Command Usage

Set Foreign Exchange Office Loop Start (FXO LS) to *detect inbound* caller ID (on/off).

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxo ls caller id {on | off}**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ Syntax Notes ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ Syntax Note ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

on — Turns ON fxo ls caller ID for specified signaling template or port.

off — Turns OFF fxo ls caller ID for specified signaling template or port.

♦ Syntax Note ♦

The **voice coding profile caller id** command must be enabled to use this command.

Default:

The default setting is **on**.

Command Examples:

voice signaling template "sigtempbranch1" fxo ls caller id on
voice signaling template "sigtempbranch2" fxo ls caller id off
voice signaling channel 2/1/1-12 fxo ls caller id on
voice signaling channel 2/2/13-24 fxo ls caller id off

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo ls answer after

Command Usage

Specify Foreign Exchange Office Loop Start (FXO LS) number of rings allowed before answering.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxo ls answer after** <*value*>

Definitions:

TemplateName	Identifies the signaling by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
	♦ **Syntax Notes** ♦ At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).
	♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
value	Specifies the number of rings permitted to elapse before answering incoming calls, per ring from 1 to 65,535, (e.g., **2**) rings.
	♦ **Syntax Note** ♦ *Do not* use commas when entering the FXS GS call originate clear detect value (for example, **10,000** will return a syntax error message).

Default:
The default value is **2** rings.

Command Examples:
voice signaling template "sigtempbranch1" 2/1 fxo ls answer after 2
voice signaling template "sigtempbranch2" 2/2 fxo ls answer after 30000
voice signaling template "sigtempbranch3" 2/3 fxo ls answer after 65535
voice signaling channel 2/1/1-12 fxo ls answer after 2
voice signaling channel 2/2/13-24 fxo ls answer after 30000
voice signaling channel 2/3/1-30 fxo ls answer after 65535

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo ls loop current debounce

Command Usage

Specify Foreign Exchange Office Loop Start (FXO LS) debounce for loop current detector.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxo ls loop current debounce <***value***>**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies the time to use as a debouncer (delay interval) for debouncing the loop current detector, in milliseconds from 0 to 60,000, (e.g., **20**).

♦ **Syntax Note** ♦

Do not use commas when entering the fxo ls debounce loop current detector value (for example, **10,000** will return a syntax error message).

Default:
The default value is **20** milliseconds.

Command Examples:
voice signaling template "sigtempbranch1" fxo ls loop current debounce 20
voice signaling template "sigtempbranch2" fxo ls loop current debounce 30000
voice signaling template "sigtempbranch3" fxo ls loop current debounce 60000
voice signaling channel 2/1/1-12 fxo ls loop current debounce 20
voice signaling channel 2/2/13-24 fxo ls loop current debounce 30000
voice signaling channel 2/3/1-30 fxo ls loop current debounce 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo ls battery reversal debounce

Command Usage

Specify Foreign Exchange Office Loop Start (FXO LS) debounce for battery reversal detector.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxo ls battery reversal debounce <***value* **>**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦
At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦
Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies the time to use as a debouncer (delay interval) for debouncing the battery reversal detector, in milliseconds from 0 to 60,000, (e.g., **20**).

♦ **Syntax Note** ♦
Do not use commas when entering the fxo ls debounce value for the battery reversal detector (for example, **10,000** will return a syntax error message).

Default:
The default value is **20** milliseconds.

Command Examples:
voice signaling template "sigtempbranch1" fxo ls battery reversal debounce 20
voice signaling template "sigtempbranch2" fxo ls battery reversal debounce 30000
voice signaling template "sigtempbranch3" fxo ls battery reversal debounce 60000
voice signaling channel 2/1/1-12 fxo ls battery reversal debounce 20
voice signaling channel 2/2/13-24 fxo ls battery reversal debounce 30000
voice signaling channel 2/3/1-30 fxo ls battery reversal debounce 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs gs seize detect

Command Usage

Specify Foreign Exchange Station Ground Start (FXO GS) time to wait before declaring off-hook.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxs gs seize detect** <*value*>

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ Syntax Notes ♦

At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ Syntax Note ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies the time to wait before declaring on-hook (seize detect) condition, in milliseconds from 0 to 5,000, (e.g., **150**).

♦ Syntax Note ♦

Do not use commas when entering the FXS GS time to wait before declaring seize detect value (for example, **2,500** will return a syntax error message).

Default:
The default value is **150** milliseconds.

Command Examples:
voice signaling template "sigtempbranch1" fxs gs seize detect 150
voice signaling template "sigtempbranch2" fxs gs seize detect 2500
voice signaling template "sigtempbranch3" fxs gs seize detect 5000
voice signaling channel 2/1/1-12 fxs gs seize detect 150
voice signaling channel 2/2/13-24 fxs gs seize detect 2500
voice signaling channel 2/3/1-30 fxs gs seize detect 5000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs gs on hook debounce

Command Usage

Specify Foreign Exchange Station Ground Start (FXS GS) debounce interval for on-hook transition.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxs gs on hook debounce <***value***>**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦
At least one ASCII character must be used in the signaling template name, and quotes must be located at each end of the name.

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦
Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies the debounce (delay interval) transition time to on-hook (seize detect) condition, in milliseconds from 0 to 1,000, (e.g., **20**).

♦ **Syntax Note** ♦
Do not use commas when entering the FXS GS debounce interval on-hook transition value (for example, **1,000** will return a syntax error message).

Default:
The default value is **20** milliseconds.

Command Examples:

```
voice signaling template "sigtempbranch1" fxs gs on hook debounce 20
voice signaling template "sigtempbranch2" fxs gs on hook debounce 500
voice signaling template "sigtempbranch3" fxs gs on hook debounce 1000
voice signaling channel 2/1/1-12 fxs gs on hook debounce 20
voice signaling channel 2/2/13-24 fxs gs on hook debounce 500
voice signaling channel 2/3/1-30 fxs gs on hook debounce 1000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs gs originate clear detect

Command Usage

Specify Foreign Exchange Station Ground Start (FXS GS) minimum time to wait before declaring on-hook by *originator.*

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxs gs originate clear detect <***value* **>**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value	Specifies the minimum time to wait if call originator hangs up before on-hook condition declared, in milliseconds from 1 to 60,000, (e.g., **200**).

♦ **Syntax Note** ♦

Do not use commas when entering the FXS GS call originate clear detect value (for example, **10,000** will return a syntax error message).

Default:
The default value is **200** milliseconds.

Command Examples:
voice signaling template "sigtempbranch1" fxs gs originate clear detect 200
voice signaling template "sigtempbranch2" fxs gs originate clear detect 30000
voice signaling template "sigtempbranch3" fxs gs originate clear detect 60000
voice signaling channel 2/1/1-12 fxs gs originate clear detect 200
voice signaling channel 2/2/13-24 fxs gs originate clear detect 30000
voice signaling channel 2/3/1-30 fxs gs originate clear detect 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs gs answer clear detect

Command Usage

Specify Foreign Exchange Station Ground Start (FXS GS) minimum time to wait before declaring on-hook by *answerer.*

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxs gs answer clear detect <***value* **>**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value	Specifies the minimum time to wait if answering party hangs up before on-hook condition declared, in milliseconds from 1 to 60,000, (e.g., **300**).

♦ **Syntax Note** ♦

Do not use commas when entering the FXS GS answering party clear detect value (for example, **1,000** will return a syntax error message).

Default:
The default value is **100** milliseconds.

Command Examples:
voice signaling template "sigtempbranch1" fxs gs answer clear detect 100
voice signaling template "sigtempbranch2" fxs gs answer clear detect 30000
voice signaling template "sigtempbranch3" fxs gs answer clear detect 60000
voice signaling channel 2/1/1-12 fxs gs answer clear detect 100
voice signaling channel 2/2/13-24 fxs gs answer clear detect 30000
voice signaling channel 2/3/1-30 fxs gs answer clear detect 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs gs min ring ground

Command Usage

Specify Foreign Exchange Station Ground Start (FXS GS) time to wait after ring before grounding tip.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxs gs min[imum] ring ground <***value***>**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value	Specifies the minimum time to wait after ring ground detection before line responds by grounding tip, in milliseconds from 0 to 65535, (e.g., **100**).
imum	Optional command syntax. You can type either **min** or **minimum** in the command line.

♦ **Syntax Note** ♦

Do not use commas when entering the minimum FXS GS time to wait value after ring ground before grounding tip (for example, **10,000** will return a syntax error message).

Default:

The default value is **100** milliseconds.

Command Examples:

voice signaling template "sigtempbranch1" fxs gs minimum ring ground 100
voice signaling template "sigtempbranch2" fxs gs min ring ground 30000
voice signaling template "sigtempbranch3" fxs gs min ring ground 65535
voice signaling channel 2/1/1-12 fxs gs minimum ring ground 100
voice signaling channel 2/2/13-24 fxs gs min ring ground 30000
voice signaling channel 2/2/1-30 fxs gs min ring ground 65535

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs gs max wait loop

Command Usage

Specify Foreign Exchange Station Ground Start (FXS GS) maximum time to wait for loop to close after grounding tip.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxs gs max[imum] wait loop** <*value*>

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

imum	Optional command syntax. You can type either **max** or **maximum** in the command line.
value	Specifies the maximum time to wait after ring ground detection for loop to close after grounding tip but before disconnecting line, in milliseconds from 0 to 65535, (e.g., **100**).

♦ **Syntax Note** ♦

Do not use commas when entering the maximum FXS GS time to wait value for loop to close after grounding tip (for example, **10,000** will return a syntax error message).

Default:
The default value is **100** milliseconds.

Command Examples:

voice signaling template "sigtempbranch1" fxs gs maximum wait loop 100
voice signaling template "sigtempbranch2" fxs gs max wait loop 30000
voice signaling template "sigtempbranch3" fxs gs max wait loop 65535
voice signaling channel 2/1/1-12 fxs gs maximum wait loop 100
voice signaling channel 2/1/13-24 fxs gs max wait loop 30000
voice signaling channel 2/1/1-30 fxs gs max wait loop 65535

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs gs min loop open

Command Usage

Specify Foreign Exchange Station Ground Start (FXS GS) minimum time between open loop and idle state.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxs gs min[imum] loop open <***value* **>**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

imum	Optional command syntax. You can type either **min** or **minimum** in the command line.
value	Specifies the maximum time to wait after ring ground detection for loop to open after grounding tip but before returning line to idle state, in milliseconds from 0 to 65535, (e.g., **100**).

♦ **Syntax Note** ♦

Do not use commas when entering the maximum FXS GS time to wait value for loop to open after grounding tip (for example, **10,000** will return a syntax error message).

Default:

The default value is **100** milliseconds.

Command Examples:

voice signaling template "sigtempbranch1" fxs gs minimum loop open 100
voice signaling template "sigtempbranch2" fxs gs min loop open 30000
voice signaling template "sigtempbranch3" fxs gs min loop open 65535
voice signaling channel 2/1/1-12 fxs gs minimum loop open 100
voice signaling channel 2/2/13-24 fxs gs min loop open 30000
voice signaling channel 2/2/1-30 fxs gs min loop open 65535

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs gs caller id

Command Usage

Set Foreign Exchange Station Ground Start (FXS GS) to *generate outbound* caller ID (on/off).

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxs gs caller id {on | off}**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

on	Turns ON FXS GS caller ID for specified signaling template or port.
off	Turns OFF FXS GS caller ID for specified signaling template or port.

♦ **Syntax Note** ♦

The **voice coding profile caller id** command must be enabled to use this command.

Default:

The default setting is **off**.

Command Examples:

voice signaling template "sigtempbranch1" 2/1 fxs gs caller id off
voice signaling template "sigtempbranch2" 2/2 fxs gs caller id on
voice signaling channel 2/1/1-12 fxs gs caller id off
voice signaling channel 2/2/13-24 fxs gs caller id on

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs gs off hook debounce

Command Usage

Specify Foreign Exchange Station Ground Start (FXS GS) debounce interval for off-hook.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxs gs off hook debounce <***value* **>**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies the debounce (delay interval) transition time to off-hook condition, in milliseconds from 0 to 1,000, (e.g., **20**).

♦ **Syntax Note** ♦

Do not use commas when entering the FXS GS debounce interval on-hook transition value (for example, **1,000** will return a syntax error message).

Default:

The default value is **20** milliseconds.

Command Examples:

voice signaling template "sigtempbranch1" fxs gs off hook debounce 20
voice signaling template "sigtempbranch2" fxs gs off hook debounce 500
voice signaling template "sigtempbranch3" fxs gs off hook debounce 1000
voice signaling channel 2/1/1-12 fxs gs off hook debounce 20
voice signaling channel 2/2/13-24 fxs gs off hook debounce 500

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs gs ring ground debounce

Command Usage

Specify Foreign Exchange Station Ground Start (FXS GS) debounce interval for ring ground detector.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxs gs ring ground debounce <***value***>**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value	Specifies how long to use the debounce (delay interval) for debouncing the ring ground detector, in milliseconds from 0 to 1,000, (e.g., **20**).

♦ **Syntax Note** ♦

Do not use commas when entering the FXS GS debounce interval value for ring ground detection (for example, **1,000** will return a syntax error message).

Default:

The default value is **20** milliseconds.

Command Examples:

voice signaling template "sigtempbranch1" fxs gs ring ground debounce 20
voice signaling template "sigtempbranch2" fxs gs ring ground debounce 500
voice signaling template "sigtempbranch3" fxs gs ring ground debounce 1000
voice signaling channel 2/1/1-12 fxs gs ring ground debounce 20
voice signaling channel 2/2/13-24 fxs gs ring ground debounce 500
voice signaling channel 2/3/1-30 fxs gs ring ground debounce 1000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs gs cadence coefficient

Command Usage

Set Foreign Exchange Station Ground Start (FXS GS) cadence coefficient, or continental ring tone (North America/Europe).

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxs gs cadence coefficient {north america | europe}**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ` ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦
Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

north america — Turns ON FXS GS cadence coefficient for North America.

europe — Turns ON FXS GS cadence coefficient for Europe.

♦ **Syntax Note** ♦
This command must be set in relation to the **voice signaling fxs gs ring ID** command.

Default:
The default setting is **north america**.

Command Examples:
voice signaling template "sigtempbranch1" fxs gs cadence coefficient north america
voice signaling template "sigtempbranch2" fxs gs cadence coefficient europe
voice signaling channel 2/1/1-12 fxs gs cadence coefficient north america
voice signaling channel 2/2/13-24 fxs gs cadence coefficient europe

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs gs ring id

Command Usage

Set Foreign Exchange Station Ground Start (FXS GS) ring ID, or continental ring tone variance for North America or Europe.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxs gs ring id {0 | 1 | 2 | 3 | 4 | 5 | 6 | 7 | default}**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦
Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

0, 1, 2, 3, 4, 5, 6, 7	Specifies variance in continental ring tone for either North America or Europe (e.g., **2**) as per selected coefficient. The value **0** is the same as **default**.
default	Sets the tone variance to the default setting, i.e, 0.

♦ **Syntax Note** ♦
This command must be set in relation to the **voice signaling fxs gs ring cadence coefficient** command.

Default:
The default value is **0**.

Command Examples:

voice signaling template "sigtempbranch1" fxs ls ring ID default
voice signaling template "sigtempbranch2" fxs ls ring ID 0
voice signaling template "sigtempbranch3" fxs ls ring ID 7
voice signaling channel 2/1/1-12 fxs gs ring ID default
voice signaling channel 2/2/13-24 fxs gs ring ID 0
voice signaling channel 2/3/1-30 fxs gs ring ID 7

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

The ring cadence that is set for the first channel on the first port is automatically set for all remaining FXS/FXO ports.

All ports must have the same ring cadence.

On the Omni Access 512, the ring cadence can only be specified for port 1.

On the Omni Switch/Router, the ring cadence can only be specified for port 1 (or 9 for a dual VSA-FXS/FXS in an HSX-H).

voice signaling fxo gs connection loop open debounce

Command Usage

Specify Foreign Exchange Office Ground Start (FXS GS) debounce interval for loop open detection.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxo gs connection loop open debounce <***value* **>**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value	Specifies the debounce (delay interval) for loop open detection for existing connection, in milliseconds from 1 to 60,000, (e.g., **150**).

♦ **Syntax Note** ♦

Do not use commas when entering the FXS GS debounce interval value for loop open detection (for example, **1,000** will return a syntax error message).

Default:
The default value is **150** milliseconds.

Command Examples:
voice signaling template "sigtempbranch1" fxo gs connection loop open debounce 150
voice signaling template "sigtempbranch2" fxo gs connection loop open debounce 30000
voice signaling template "sigtempbranch3" fxo gs connection loop open debounce 60000
voice signaling channel 2/1/1-12 fxo gs connection loop open debounce150
voice signaling channel 2/2/13-24 fxo gs connection loop open debounce 30000
voice signaling channel 2/3/1-30 fxo gs connection loop open debounce 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo gs max tip ground wait

Command Usage

Specify Foreign Exchange Office Ground Start (FXS GS) maximum time between ring ground and tip ground.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxo gs max[imum] tip ground wait <***value***>**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦
Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

imum	Optional command syntax. You can type either **max** or **maximum** in the command line.
value	Specifies the maximum time the line waits after ring ground asserted for tip ground received, in milliseconds from 1 to 60,000, (e.g., **150**).

♦ **Syntax Note** ♦
Do not use commas when entering the maximum FXS GS value for time between ring ground and tip ground (for example, **1,000** will return a syntax error message).

Default:
The default value is **30** milliseconds.

Command Examples:
voice signaling template "sigtempbranch1" fxo gs maximum tip ground wait 30
voice signaling template "sigtempbranch2" fxo gs max tip ground wait 30000
voice signaling template "sigtempbranch3" fxo gs max tip ground wait 60000
voice signaling channel 2/1/1-12 fxo gs maximum tip ground wait 30
voice signaling channel 2/2/13-24 fxo gs max tip ground wait 30000
voice signaling channel 2/3/1-30 fxo gs max tip ground wait 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo gs tip ground wait debounce

Command Usage

Specify Foreign Exchange Office Ground Start (FXS GS) debounce interval for tip ground detector.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxo gs tip ground wait debounce <***value***>**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value	Specifies the debounce (delay interval) for debouncing the tip ground detector, in milliseconds from 1 to 1,000, (e.g., **20**).

♦ **Syntax Note** ♦

Do not use commas when entering the FXS GS debounce interval value for loop open detection (for example, **1,000** will return a syntax error message).

Default:
The default value is **20** milliseconds.

Command Examples:

voice signaling template "sigtempbranch1" fxo gs tip ground wait debounce 20
voice signaling template "sigtempbranch2" fxo gs tip ground wait debounce 500
voice signaling template "sigtempbranch3" fxo gs tip ground wait debounce 1000
voice signaling channel 2/1/1-12 fxo gs tip ground wait debounce 20
voice signaling channel 2/2/13-24 fxo gs tip ground wait debounce 500
voice signaling channel 2/3/1-30 fxo gs tip ground wait debounce 1000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo gs ringing debounce

Command Usage

Specify Foreign Exchange Office Ground Start (FXS GS) debounce for incoming ring signal.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxo gs ringing debounce** <*value*>

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value — Specifies the debounce (delay interval) for incoming ring signal, in milliseconds from 1 to 1,000, (e.g., **50**).

♦ **Syntax Note** ♦

Do not use commas when entering the FXS GS debounce interval value for incoming ring signal (for example, **1,000** will return a syntax error message).

Default:

The default value is **50** milliseconds.

Command Examples:

voice signaling template "sigtempbranch1" fxo gs ringing debounce 50
voice signaling template "sigtempbranch2" fxo gs ringing debounce 5
voice signaling template "sigtempbranch3" fxo gs ringing debounce 1000
voice signaling channel 2/1/1-12 fxo gs ringing debounce 50
voice signaling channel 2/2/13-24 fxo gs ringing debounce 5
voice signaling channel 2/3/1-30 fxo gs ringing debounce 1000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo gs ringing inter cycle

Command Usage

Specify Foreign Exchange Office Ground Start (FXS GS) time between consecutive ring *cycles.*

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxo gs ringing inter cycle <***value***>**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**). ♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
value	Specifies the time between FXS GS start time between consecutive ring cycles, in milliseconds from 1 to 1,000, (e.g., **5000**), to detect ringing. ♦ **Syntax Note** ♦ *Do not* use commas when entering the FXS GS start time between consecutive ring cycles to detect ringing (for example, **5,000** will return a syntax error message).

Default:
The default value is **5000** milliseconds.

Command Examples:
voice signaling template "sigtempbranch1" fxo gs ringing inter cycle 5000
voice signaling template "sigtempbranch2" fxo gs ringing inter cycle 30000
voice signaling template "sigtempbranch3" fxo gs ringing inter cycle 60000
voice signaling channel 2/1/1-12 fxo gs ringing inter cycle 5000
voice signaling channel 2/2/13-24 fxo gs ringing inter cycle 30000
voice signaling channel 2/3/1-30 fxo gs ringing inter cycle 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo gs ringing inter pulse

Command Usage

Specify Foreign Exchange Office Ground Start (FXS GS) time between consecutive ring *pulses.*

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxo gs ringing inter pulse <***value***>**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value	Specifies the time between fxo gs start time between consecutive ring pulses in the same ring cycle, in milliseconds from 1 to 60,000, (e.g., **550**), to detect ringing.

♦ **Syntax Note** ♦

Do not use commas when entering the FXS GS start time between consecutive ring pulses to detect ringing (for example, **5,000** will return a syntax error message).

Default:

The default value is **550** milliseconds.

Command Examples:

voice signaling template "sigtempbranch1" fxo gs ringing inter pulse 550
voice signaling template "sigtempbranch2" fxo gs ringing inter pulse 30000
voice signaling template "sigtempbranch3" fxo gs ringing inter pulse 60000
voice signaling channel 2/1/1-12 fxo gs ringing inter pulse 550
voice signaling channel 2/2/13-24 fxo gs ringing inter pulse 30000
voice signaling channel 2/3/1-30 fxo gs ringing inter pulse 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo gs caller id detection

Command Usage

Set Foreign Exchange Office (FXO) Ground Start (GS) to *detect inbound* caller ID (on/off).

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxo gs caller id detection {on | off}**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

on	Turns ON FXS GS caller ID detection for specified signaling template or port.
off	Turns OFF FXS GS caller ID detection for specified signaling template or port.

♦ **Syntax Note** ♦

The **voice coding profile caller id** command must be enabled to use this command.

Default:

The default setting is **off**.

Command Examples:

voice signaling template "sigtempbranch1" fxo gs caller id detection off
voice signaling template "sigtempbranch2" fxo gs caller id detection on
voice signaling channel 2/1/1-12 fxo gs caller id detection off
voice signaling channel 2/2/13-24 fxo gs caller id detection on

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo gs answer after

Command Usage

Specify Foreign Exchange Office (FXO) Ground Start (GS) number of rings allowed before answering.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxo gs answer after** <*value*>

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**). ♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
value	Specifies the number of rings permitted to elapse before answering incoming calls, per ring from 1 to 65,535, (e.g., **2**) rings. ♦ **Syntax Note** ♦ *Do not* use commas when entering the FXS GS call answer after rings value (for example, **10,000** will return a syntax error message). If caller ID is ON, then the number of rings should be greater than or equal to 2 rings.

Default:
The default value is **50** rings.

Command Examples:
voice signaling template "sigtempbranch1" 2/1 fxo gs answer after 50
voice signaling template "sigtempbranch2" 2/2 fxo gs answer after 5
voice signaling template "sigtempbranch3" 2/3 fxo gs answer after 1000
voice signaling channel 2/1/1-12 fxo gs answer after 50
voice signaling channel 2/2/13-24 fxo gs answer after 5
voice signaling channel 2/3/1-30 fxo gs answer after 1000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo gs loop current debounce

Command Usage

Specify Foreign Exchange Office (FXO) Ground Start (GS) debounce interval for loop current detector.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} fxo gs loop current debounce <***value***>**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value	Specifies the debounce (delay interval) for debouncing the loop current detector, in milliseconds from 1 to 1,000, (e.g., **20**).

♦ **Syntax Note** ♦

Do not use commas when entering the FXS GS debounce interval value for loop current detection (for example, **1,000** will return a syntax error message).

Default:

The default value is **20** milliseconds.

Command Examples:

voice signaling template "sigtempbranch1" fxo gs loop current debounce 20
voice signaling template "sigtempbranch2" fxo gs loop current debounce 500
voice signaling template "sigtempbranch3" fxo gs loop current debounce 1000
voice signaling channel 2/1/1-12 fxo gs loop current debounce 20
voice signaling channel 2/2/13-24 fxo gs loop current debounce 500
voice signaling channel 2/3/1-30 fxo gs loop current debounce 1000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo gs battery reversal debounce

Command Usage

Specify Foreign Exchange Office (FXO) Ground Start (GS) debounce interval for battery reversal detector.

Syntax Options

voice signaling {template ***"TemplateName"*** **| channel** ***slot/port/startChannel-endChannel*** **} fxo gs battery reversal debounce <*value*>**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & I / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

valueS	Specifies the time to use as a debouncer (delay interval) for debouncing the battery reversal detector, in milliseconds from 1 to 1,000, (e.g., **20**).

♦ **Syntax Note** ♦

Do not use commas when entering the FXS GS debounce value for the battery reversal detector (for example, **1,000** will return a syntax error message).

Default:

The default value is **20** milliseconds.

Command Examples:

voice signaling template "sigtempbranch1" fxo gs battery reversal debounce 20
voice signaling template "sigtempbranch2" fxo gs battery reversal debounce 500
voice signaling template "sigtempbranch3" fxo gs battery reversal debounce 1000
voice signaling channel 2/1/1-12 fxo gs battery reversal debounce 20
voice signaling channel 2/2/13-24 fxo gs battery reversal debounce 500
voice signaling channel 2/3/1-30 fxo gs battery reversal debounce 1000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling caller id name

Command Usage

Set outbound caller ID name (private/unavailable) to transmit.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} caller id name {** *"callerIdName"* **| private | unavailable}**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ Syntax Note ♦
Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

callerIdName	Identifies the caller by name, (e.g., **salemcaller**); maximum length of 15 characters. The following characters are permitted in the caller ID name: a-z, A-Z, _ , and no spaces are allowed.
private	Sets outbound originating caller ID name that transmits to "private" (may be abbreviated to "p").
unavailable	Sets outbound originating caller ID name that transmits to "unavailable" (may be abbreviate to "o").

♦ Syntax Note ♦
The **voice coding profile caller id** command must be enabled to use this command.

Default:
The default setting is **private**.

Command Examples:
voice signaling template "sigtempbranch1" "salemcaller" private
voice signaling template "sigtempbranch2" "calabcaller" p
voice signaling template "sigtempbranch3" "salemcaller" unavailable
voice signaling template "sigtempbranch4" "calabcaller" o
voice signaling channel 2/1/1-12 "salemcaller" private
voice signaling channel 2/2/13-24 "calabcaller" p

Remarks

Caller ID time is automatically determined (read) from the system time on the switch. If the caller ID name or number is changed, the voice switching daughtercard is initialized with the time automatically.

voice signaling caller id number

Command Usage

Set outbound caller ID number (published/non-published) number to transmit.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} caller id number {** *"callerIdNumber"* **| private | unavailable}**

Definitions:

TemplateName	Identifies the telephony signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦
Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

callerIdNumber	Identifies the caller by number, (e.g., **8188803500**); maximum length of ten characters (only 0-9 allowed); no _ (underscores) or spaces allowed.
on	Sets outbound originating caller ID number that transmits to "private" (may be abbreviated to "p").
off	Sets outbound originating caller ID number that transmits to "unavailable" (may be abbreviated to "o").

♦ **Syntax Note** ♦
The **voice coding profile caller id** command must be enabled to use this command.

Default:
The default setting is **private**.

Command Examples:
voice signaling template "sigtempbranch1" caller id number "8188803500" private
voice signaling template "sigtempbranch2" caller id number "8188803501" p
voice signaling template "sigtempbranch3" caller id number "8188803502" unavailable
voice signaling template "sigtempbramch4" caller id number "8188803503" o
voice signaling channel 2/1/1-12 caller id number "8188803500" private
voice signaling channel 2/2/13-24 caller id number 8188803501" unavailable

Remarks

Caller ID time is automatically determined (read) from the system time on the switch. If the caller ID name or number is changed, the voice switching daughtercard is initialized with the time automatically.

voice signaling tone table

Command Usage

Set outbound tone table (ringing/silence) for telephony channel identifier (TCID).

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} tone table {ringing | silence}**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**). Consists of at least one ASCII character with quotes on each end of the name; maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ &	/ \ < > ( ) [ ] { }
slot	Specifies slot number of voice daughtercard installed in switching module, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

ringing	Sets tone table for TCID to ringing (for "normal" ringback sound).
silence	Sets tone table for TCID to silence (for "silent" ringback).

Default:

The default setting is **ringing**.

Command Examples:

voice signaling template "sig tempbranch1" tone table ringing
voice signaling template sigtempbranch2 tone table silence
voice signaling channel 2/1/1-12 tone table ringing
voice signaling channel 2/2/13-24 tone table silence

voice signaling call progress tones

Command Usage

Set call signaling for detection of call progress tones (on/off/relative).

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} call progress tones {on | off | relative}**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ Syntax Note ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

on	Turns ON call progress tone detection for specified signaling template or port.
off	Turns OFF call progress tones detection for specified signaling template or port.
relative	Turns call progress tone detection ON or OFF according to the "call progress tone detection" parameter in the currently loaded coding profile.

Default:

The default setting is **off**.

Command Examples:

voice signaling template "sigtempbranch1" call progress tones off
voice signaling template "sigtempbranch2" call progress tones on
voice signaling channel 2/1/1-12 call progress tones off
voice signaling channel 2/2/13-24 call progress tones on

voice signaling call progress tone detection configuration

Command Usage

Set call signaling for call progress tone detection configuration (default/alternate).

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} call progress tone detection configuration {default | alternate}**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).
	♦ Syntax Note ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
default	Use default call progress tone detection configuration for specified signaling template or port.
alternate	Use alternate call progress tone detection configuration for specified signaling template or port.

Default:
The default setting is **default**.

Command Examples:
voice signaling template "sigtempbranch1" call progress tone detection configuration default
voice signaling template "sigtempbranch2" call progress tone detection configuration alternate
voice signaling template channel 2/1/1-12 call progress tone detection configuration default
voice signaling template channel 2/2/13-24 call progress tone detection configuration alternate

Remarks

This command is used to specify which configuration to use for call progress tone detection. Each configuration contains filter configuration information (threshold and filter coefficients), and a table containing cadence information of all the call progress tones that need to be detected. Each configuration, whether default or alternate has a filter configuration for dial tone, ring back, including three supported cadences, bust, and congestion.

If the alternative tone detection configuration is selected, the tone detection process is limited to a busy tone, and other detected tones are ignored.

voice signaling v.18 tone detection threshold hang time

Command Usage

Set V.18 Annex A signal duration threshold for tone detection.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} v.18 tone detection threshold hang time <***value* **>**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & ! / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value	Specifies the signal duration threshold for V.18 tone detection, in milliseconds from 5 to 32767, (e.g., **20**).

♦ **Syntax Note** ♦

Do not use commas when entering the V.18 tone detection threshold hang time value (for example, **1,000** will return a syntax error message).

Default:
The default value is **20** milliseconds.

Command Examples:
voice signaling template "sigtempbranch1" v.18 tone detection threshold hang time 50
voice signaling template "sigtempbranch2" v.18 tone detection threshold hang time 10000
voice signaling template "sigtempbranch1" v. 18 tone detection threshold hang time 32767
voice signaling template channel 2/1/1-12 v.18 tone detection threshold hang time 50
voice signaling template channel 2/2/13-24 v.18 tone detection threshold hang time 10000
voice signaling template channel 2/3/1-30 v.18 tone detection threshold hang time 32767

Remarks

V.18 threshold commands are used to specify minimum detectable signal values, and are especially geared toward improving telecommunications for the hearing or speech-impaired; for example, various thresholds are available to control duration (hang time) and strength (level) or even incremental (fractional) signaling.

voice signaling v.18 tone detection threshold level

Command Usage

Set V.18 Annex A signal strength threshold level for tone detection.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} v.18 tone detection threshold level <***value***>**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦
Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value	Specifies the signal duration threshold for V.18 tone detection, in dBm0 (decibels below 0 milliwatts) for output with no input power, from **-50** to **-15,** (e.g., **-50**).

Default:
The default value is **-40** dBm0.

Command Examples:
voice signaling template "sigtempbranch2" v.18 tone detection threshold level -15
voice signaling template channel 2/2/13-24 v.18 tone detection threshold level -15

Remarks

V.18 threshold commands are used to specify minimum detectable signal values, and are especially geared toward improving telecommunications for the hearing or speech-impaired; for example, various thresholds are available to control duration (hang time) and strength (level) or even incremental (fractional) signaling.

voice signaling v.18 tone detection threshold fraction

Command Usage

Set V.18 Annex A signal strength threshold fraction for tone detection.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} v.18 tone detection threshold fraction <***value***>**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value	Specifies the signal fraction threshold for V.18 tone detection, in dBm0 (decibels below 0 milliwatt; output with no input power), 1 to 32767, (e.g., **10**).

♦ **Syntax Note** ♦

Do not use commas when entering the V.18 tone detection threshold fraction value (for example, **1,000** will return a syntax error message).

Default:
The default value is **10** dBm0.

Command Examples:
voice signaling template "sigtempbranch1" v.18 tone detection threshold fraction 10
voice signaling template "sigtempbranch2" v.18 tone detection threshold fraction 10000
voice signaling template "sigtempbranch1" v. 18 tone detection threshold fraction 32767
voice signaling template channel 2/1/1-12 v.18 tone detection threshold fraction 10
voice signaling template channel 2/2/13-24 v.18 tone detection threshold fraction 10000
voice signaling template channel 2/3/1-30 v.18 tone detection threshold fraction 32767

Remarks

V.18 threshold commands are used to specify minimum detectable signal values, and are especially geared toward improving telecommunications for the hearing or speech-impaired; for example, various thresholds are available to control duration (hang time) and strength (level) or even incremental (fractional) signaling.

voice signaling single frequency tone detection level

Command Usage

Set signal strength threshold level for single (frequency) tone detection.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} single frequency tone detection level <***value* **>**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ` ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**). ♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
value	Specifies the signal frequency threshold level for tone detection, in dBm (decibels below 1 milliwatt; output signal power referenced to 1 milliwatt input signal power), from **-50** to **-15**, (e.g., **-40**).

Default:
The default value is **-40** dBm.

Command Examples:
voice signaling template "sigtempbranch1" single frequency tone detection threshold level -40
voice signaling template "sigtempbranch2" single frequency tone detection threshold level -50
voice signaling template channel 2/1/1-12 single frequency tone detection threshold level -40
voice signaling template channel 2/2/13-24 single frequency tone detection threshold level -50

voice signaling single frequency tone detection threshold time

Command Usage

Set signal strength threshold duration for single (frequency) tone detection.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} single frequency tone detection threshold time <***value***>**

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**). **♦ Syntax Note ♦** Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
value	Specifies the signal threshold time for single frequency tone detection, in milliseconds from 0 to 65535, (e.g., **50**). **♦ Syntax Note ♦** *Do not* use commas when entering the single frequency tone detection threshold time (for example, **20,000** will return a syntax error message).

Default:
The default value is **50** milliseconds.

Command Examples:
voice signaling template "sigtempbranch1" single frequency tone detection threshold time 50
voice signaling template "sigtempbranch2" single frequency tone detection threshold time 20000
voice signaling template "sigtempbranch1" single frequency tone detection threshold time 655357
voice signaling template channel 2/1/1-12 single frequency tone detection threshold time 50
voice signaling template channel 2/2/13-24 single frequency tone detection threshold time 20000
voice signaling template channel 2/3/1-30 single frequency tone detection threshold time 65535

voice signaling echo canceller non-linear sensitivity

Command Usage

Set echo canceller processor non-linear signal sensitivity.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} echo canceller non-linear sensitivity <***value***>**

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**). ♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
value	Specifies the echo canceller non-linear sensitivity, in milliseconds from 0 to 32767, (e.g., **327**). ♦ **Syntax Note** ♦ *Do not* use commas when entering the echo canceller non-linear sensitivity value (for example, **20,000** will return a syntax error message).

Default:
The default value is **327** milliseconds.

Command Example:
voice signaling template "sigtempbranch1" echo canceller non-linear sensitivity 327
voice signaling template "sigtempbranch2" echo canceller non-linear sensitivity 20000
voice signaling template "sigtempbranch1" echo canceller non-linear sensitivity 655357
voice signaling template channel 2/1/1-12 echo canceller non-linear sensitivity 327
voice signaling template channel 2/2/13-24 echo canceller non-linear sensitivity 20000
voice signaling template channel 2/3/1-30 echo canceller non-linear sensitivity 65535

Remarks

Echo cancellers handle signal transmission echoes by isolating and filtering signals. Non-linear sensitivity echo cancellers are used to adjust the output and input amplitudes of a signal, and function as comfort noise generators.

voice signaling acoustic echo canceller mode

Command Usage

Set acoustic echo canceller processor mode (on/off).

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} acoustic echo canceller mode {on | off}**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦
Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

on	Turns ON acoustic echo canceller at channel startup for specified signaling template or port.
off	Turns OFF acoustic echo canceller at channel startup for specified signaling template or port.

Default:
The default setting is **off**.

Command Example:
voice signaling template "sigtempbranch1" acoustic echo canceller mode off
voice signaling template "sigtempbranch2" acoustic echo canceller mode on
voice signaling template channel 2/1/1-12 acoustic echo canceller mode off
voice signaling template channel 2/2/13-24 acoustic echo canceller mode on

Remarks

Acoustic echo cancellers handle signal transmission echoes, on calls originating from or being sent to IP telephones, by isolating and filtering the signals. Acoustic echo cancellers function as comfort noise generators.

voice signaling acoustic echo canceller non-linear processor

Command Usage

Set acoustic echo canceller non-linear processor mode (on/off).

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} acoustic echo canceller non-linear processor {on | off}**

Definitions:

TemplateName	Identifies the signaling template by name, (e g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦
Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

on	Turns ON non-linear acoustic echo canceller for specified signaling template or port.
off	Turns OFF non-linear acoustic echo canceller for specified signaling template or port.

♦ **Syntax Note** ♦
To use this command, the acoustic echo canceller must be enabled via the **voice signaling acoustic echo canceller non-linear processor** command.

Default:
The default setting is **off**.

Command Example:
voice signaling template "sigtempbranch1" acoustic echo canceller non-linear processor off
voice signaling template "sigtempbranch2" acoustic echo canceller non-linear processor on
voice signaling template channel 2/1/1-12 acoustic echo canceller non-linear processor off
voice signaling template channel 2/2/13-24 acoustic echo canceller non-linear processor on

Remarks

Acoustic echo cancellers handle signal transmission echoes, on calls originating from or being sent to IP telephones, by isolating and filtering the signals. Acoustic echo cancellers function as comfort noise generators.

voice signaling acoustic echo canceller output

Command Usage

Set acoustic echo canceller processor output (handset/handsfree).

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} acoustic echo canceller output {handset | handsfree}**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ Syntax Note ♦
Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

handset	Set acoustic echo canceller output to handset for specified signaling template or port.
handsfree	Set acoustic echo canceller output to handsfree for specified signaling template or port.

♦ Syntax Note ♦
To use this command, the acoustic echo canceller must be enabled via the **voice signaling acoustic echo canceller non-linear processor** command.

Default:
The default setting is **handsfree**.

Command Example:
voice signaling template "sigtempbranch1" acoustic echo canceller output handset
voice signaling template "sigtempbranch2" acoustic echo canceller output handsfree
voice signaling template channel 2/1/1-12 acoustic echo canceller output handset
voice signaling template channel 2/2/13-24 acoustic echo canceller output handsfree

Remarks

Acoustic echo cancellers handle signal transmission echoes, on calls originating from or being sent to IP telephones, by isolating and filtering the signals. Acoustic echo cancellers function as comfort noise generators.

voice signaling acoustic echo canceller handset speaker gain

Command Usage

Set acoustic voice echo canceller processor handset speaker gain.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel***} acoustic echo canceller handset speaker gain <***value***>**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value	Specifies the acoustic echo canceller handset speaker gain, in milliseconds from 0 to 31, (e.g.,**10**).

♦ **Syntax Note** ♦

To use this command, the acoustic echo canceller must be enabled via the **voice signaling acoustic echo canceller non-linear processor** command, and the **acoustic echo canceller output** command must be set to handset.

Default:
The default value is **10** milliseconds.

Command Example:
voice signaling template "sigtempbranch1" acoustic echo canceller handset speaker gain 10
voice signaling template "sigtempbranch2" acoustic echo canceller handset speaker gain 31
voice signaling template channel 2/1/1-12 acoustic echo canceller handset speaker gain 10
voice signaling template channel 2/2/13-24 acoustic echo canceller handset speaker gain 31

Remarks

Acoustic echo cancellers handle signal transmission echoes, on calls originating from or being sent to IP telephones, by isolating and filtering the signals. Acoustic echo cancellers function as comfort noise generators.

voice signaling acoustic echo canceller handsfree speaker gain

Command Usage

Set acoustic echo canceller handsfree speaker gain.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} acoustic echo canceller handsfree speaker gain** <*value*>

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).
	♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
value	Specifies the acoustic echo canceller handsfree speaker gain, in milliseconds from 0 to 31, (e.g., **10**).
	♦ **Syntax Note** ♦ To use this command, the acoustic echo canceller must be enabled via the **voice signaling acoustic echo canceller non-linear processor** command, and the **acoustic echo canceller output** command must be set to handsfree.

Default:
The default value is **10** milliseconds.

Command Example:
voice signaling template "sigtempbranch1" acoustic echo canceller handsfree speaker gain 10
voice signaling template "sigtempbranch2" acoustic echo canceller handsfree speaker gain 31
voice signaling template channel 2/1/1-12 acoustic echo canceller handsfree speaker gain 10
voice signaling template channel 2/2/13-24 acoustic echo canceller handsfree speaker gain 31

Remarks

Acoustic echo cancellers handle signal transmission echoes, on calls originating from or being sent to IP telephones, by isolating and filtering the signals. Acoustic echo cancellers function as comfort noise generators.

voice signaling override in band call progress tones

Command Usage

Override call signaling for detection of call progress tones (on/off/relative). This command should only be used under the supervision of trained personnel.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} [no] override in band call progress tones {on | off}**

Definitions:

TemplateName — Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

slot — Specifies chassis slot number where VSM is installed, (e.g., **2**).

port — Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel — The first number in the range of voice channels (e.g., **1**).

endChannel — The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

no — Optional command syntax.

♦ **Syntax Note** ♦

If **no** is used then ON and OFF cannot be specified.

on — Turns ON passing of in band call progress tones as voice data (after net-connect and before tele-connect state) for specified signaling template or port.

off — Turns OFF passing of in band call progress tone as voice data (after net-connect and before tele-connect state) for specified signaling template or port.

Default:

The default setting is **no override in band call progress tones.**

Command Examples:

voice signaling template "sigtempbranch1" no override in band call progress tones
voice signaling template "sigtempbranch2" override in band call progress off
voice signaling channel 2/1/1-12 no override in band call progress tones
voice signaling channel 2/2/13-24 override in band call progress tones on

voice signaling override full call progress tones

Command Usage

Override call signaling for call progress tone detection configuration (default/alternate). This command should only be used under the supervision of trained personnel.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} [no] override full call progress tones [on | off]**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**). ♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
no	Optional command syntax. ♦ **Syntax Note** ♦ If **no** is used then ON and OFF cannot be specified.
on	Use full call progress tones for specified signaling template or port.
off	Use full call progress tones for specified signaling template or port.

Default:
The default setting is **no override full call progress tones.**

Command Examples:
voice signaling template "sigtempbranch1" no override full call progress tones
voice signaling template "sigtempbranch2" override full call progress tones on

Remarks

This command is used to specify which configuration to use for call progress tone detection. Each configuration contains filter information (threshold and filter coefficients), and a table containing cadence information of all the call progress tones that need to be detected. The information (call progress tones) is transferred as information type packets to a coding profile. Full call progress tones means that all available call progress tones can be used in band. Each configuration, whether default or alternate has a filter configuration for dial tone, ring back, including three supported cadences, bust, and congestion.

If the alternative tone detection configuration is selected, the tone detection process is limited to a busy tone, and other detected tones are ignored.

voice signaling override ring back

Command Usage

Override call signaling for ring back (on/off). This command should only be used under the supervision of trained personnel.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} [no] override ring back {on | off}**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).
	♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
no	Optional command syntax.
	♦ **Syntax Note** ♦ If **no** is used then ON and OFF cannot be specified.
on	Turns ON ring back for specified signaling template or port.
off	Turns OFF ring back for specified signaling template or port.
	♦ **Syntax Note** ♦ To use this command, call signaling must be turned OFF via the **voice signaling override full call progress tones** command.

Default:
The default setting is **no override ring back.**

Command Examples:
voice signaling template "sigtempbranch1" no override ring back
voice signaling template "sigtempbranch2" override ring back on
voice signaling template "sigtempbranch3" override ring back off
voice signaling channel 2/1/1-12 no override ring back
voice signaling channel 2/2/13-24 override ring back on
voice signaling channel 2/3/1-30 override ring back off

Remarks

Ring back is the only call progress indication supported in the signaling band.

voice signaling override in band codec switching

Command Usage

Override call signaling for in-band codec switching (on/off). This command should only be used under the supervision of trained personnel.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} [no] override in band codec switching {on | off}**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ` ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**). ♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
no	Optional command syntax. ♦ **Syntax Note** ♦ If **no** is used then ON and OFF cannot be specified.
on	Turns ON in band codec switching for specified signaling template or port.
off	Turns OFF in band codec switching for specified signaling template or port. ♦ **Syntax Note** ♦ To use this command, call signaling must be set to voice via the call signaling **voice, fax, modem, data setup** command.

Default:
The default setting is **no override in band codec switching.**

Command Examples:
voice signaling template "sigtempbranch1" no override in band codec switching
voice signaling template "sigtempbranch2" override in band codec switching on
voice signaling template "sigtempbranch3" override in band codec switching off
voice signaling channel 2/1/1-12 no override in band codec switching
voice signaling channel 2/2/13-24 override in band codec switching on
voice signaling channel 2/3/1-30 override in band codec switching off

Remarks

In voice mode, this command enables switching from one in band codec to another by detecting changes in the payload packet type.

voice signaling override psu codec switching

Command Usage

Override call signaling for packet switch unit (PSU) codec switching (on/off). This command should only be used under the supervision of trained personnel.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} [no] override psu codec switching {on | off}**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**). ♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
no	Optional command syntax. ♦ **Syntax Note** ♦ If **no** is used then ON and OFF cannot be specified.
on	Turns ON PSU codec switching (after net-connect and before tele-connect state) for specified signaling template or port.
off	Turns OFF PSU codec switching (after net-connect and before tele-connect state) for specified signaling template or port.

Default:
The default setting is **no override psu codec switching.**

Command Examples:
voice signaling template "sigtempbranch1" no override psu codec switching
voice signaling template "sigtempbranch2" override psu codec switching on
voice signaling template "sigtempbranch3" override psu codec switching off
voice signaling channel 2/1/1-12 no override psu codec switching
voice signaling channel 2/2/13-24 override psu codec switching on
voice signaling channel 2/2/1-30 override psu codec switching off

voice signaling override network overlap dialing

Command Usage

Override call signaling for network overlap dialing (on/off). This command should only be used under the supervision of trained personnel.

Syntax Options

voice signaling {template ***"TemplateName"*** **| channel** ***slot/port/startChannel-endChannel*** **} [no] override network overlap dialing {on | off}**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦
Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

no	Optional command syntax.

♦ **Syntax Note** ♦
If **no** is used then ON and OFF cannot be specified.

on	Turns ON network overlap dialing for specified signaling template or port.
off	Turns OFF network overlap dialing for specified signaling template or port.

Default:
The default setting is **no override network overlap dialing.**

Command Examples:
voice signaling template "sigtempbranch1" no override network overlap dialing
voice signaling template "sigtempbranch2" override network overlap dialing on
voice signaling template "sigtempbranch3" override network overlap dialing off
voice signaling channel 2/1/1-12 no override network overlap dialing
voice signaling channel 2/2/13-24 override network overlap dialing on
voice signaling channel 2/3/1-30 override network overlap dialing off

Remarks

The call moves to call progress when dialing is completed, and net-connect occurs whether or not all digits have been collected.

voice signaling override information element transport

Command Usage

Override call signaling for information element (IE) transport (on/off). This command should only be used under the supervision of trained personnel.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} [no] override information element transport {on | off}**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ Syntax Note ♦
Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

no	Optional command syntax.

♦ Syntax Note ♦
If **no** is used then ON and OFF cannot be specified.

on	Turns ON information element transport for specified signaling template or port.
off	Turns OFF information element transport for specified signaling template or port.

♦ Syntax Notes ♦
If this command is set to ON, then call signaling for QSIG information must be set to OFF via the **override call signaling for qsig ie transport** command.

Default:
The default setting is **no override information element transport**.

Command Examples:
voice signaling template "sigtempbranch1" no override information element transport
voice signaling template "sigtempbranch2" override information element transport on
voice signaling template "sigtempbranch3" override information element transport off
voice signaling channel 2/1/1-12 no override information element transport
voice signaling channel 2/2/13-24 override information element transport on
voice signaling channel 2/3/1-30 override information element transport off

Remarks

This command controls transport of general user-to-user Information Element (IE) packets containing data fields of information.

voice signaling override qsig information element transport

Command Usage

Override call signaling for QSIG information element (IE) transport (on/off). This command should only be used under the supervision of trained personnel.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} [no] override qsig information element transport {on | off}**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).
	♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
no	Optional command syntax.
	♦ **Syntax Note** ♦ If **no** is used then ON and OFF cannot be specified.
on	Turns ON QSIG information element transport for specified signaling template or port.
off	Turns OFF QSIG information element transport for specified signaling template or port.
	♦ **Syntax Notes** ♦ If this command is set to ON, then call signaling for the Information Element must be set to OFF via the **override call signaling for ie transport** command.

Default:
The default setting is **no override qsig information element transport**.

Command Examples:
voice signaling template "sigtempbranch1" no override qsig information element transport
voice signaling template "sigtempbranch2" override qsig information element transport on
voice signaling template "sigtempbranch3" override qsig information element transport off
voice signaling channel 2/1/1-12 no override qsig information element transport
voice signaling channel 2/2/13-24 override qsig information element transport on
voice signaling channel 2/3/1-30 override qsig information element transport off

voice signaling override setup

Command Usage

Override call signaling for voice, fax, modem, data setup (on/off). This command should only be used under the supervision of trained personnel.

Syntax Options

voice signaling {template *"TemplateName"* **| channel** *slot/port/startChannel-endChannel* **} [no] override {voice setup | fax setup | modem setup | data setup} {on | off}**

Definitions:

TemplateName	Identifies the signaling template by name, (e.g., **sigtempbranch1**); maximum length of 40 characters. The following characters are permitted in the signaling template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ Syntax Note ♦
Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

no	Optional command syntax.

♦ Syntax Note ♦
If **no** is used then ON and OFF cannot be specified.

on	Turns ON voice, fax (hard-coded), modem or data call setup for specified signaling template or port.
off	Turns OFF voice, fax (hard-coded), modem or data call setup for specified signaling template or port.

Default:
The default setting is **no override data setup.**

Command Examples:

voice signaling template "sigtempbranch1" no override voice setup
voice signaling template "sigtempbranch2" override voice setup on
voice signaling template "sigtempbranch3" override voice setup off
voice signaling template "sigtempbranch3" no override fax setup
voice signaling template "sigtempbranch4" override fax setup on
voice signaling template "sigtempbranch5" override fax setup off
voice signaling channel 2/1/1-12 no override modem setup
voice signaling channel 2/2/13-24 override modem setup on
voice signaling channel 2/3/1-30 override modem setup off
voice signaling channel 2/1/1-12 no override data setup
voice signaling channel 2/2/13-24 override data setup on
voice signaling channel 2/3/1-30 override data setup off

Coding Profiles

The commands listed and described below are used to relate Coding Profiles to channels and configure the following associated components and functions: codecs, caller ID, voice mode parameters, voice network buffers, voice activity detection, tone detection, echo canceller, facsimile modem, facsimile T.38 mode, and silence detection.

Create Coding Profile

Delete Coding Profile

View Coding Profile

Reset all Coding Profiles to Factory Defaults

Relate to Channels

voice channel coding profile

preferred coding profile (voice, fax, modem, data) for calls on specified voice channel (optional)

General Parameters

codec type for coding profile

Voice Mode Parameters

coding profile voice packet interval size and field information size

Voice Network Buffer

coding profile buffer mode (adaptive/static)

coding profile nominal delay buffer

coding profile maximum delay buffer

Voice Activity Detector (VAD)

coding profile voice activity detector (on/off)

coding profile VAD threshold mode (adaptive/relative)

coding profile VAD audio threshold level (adaptive/relative; adaptive if threshold mode relative)

Tone Detection

coding profile voice DTMF relay (on/off)

coding profile fax modem switchover (enable/disable)

coding profile call progress tone detection (on/off)

coding profile V.18 Annex A call progress tone detection (on/off)

coding profile single frequency tone detection (on/off)

Echo Canceller

coding profile voice echo canceller (on/off)

coding profile voice echo canceller non-linear processor mode (on/off)

coding profile voice echo canceller comfort noise mode (static/mode)

coding profile voice echo canceller noise level (dBm)

coding profile voice echo canceller tail delay length

coding profile voice echo canceller refresh configuration state (refresh/freeze)

coding profile voice echo canceller coefficient refresh state (on/off)

Facsimile Modem

coding profile maximum allowed fax/modem data rate

coding profile fax/modem transmit level gain

coding profile fax/modem carrier detect threshold

coding profile inactivity detection time to automatically tear down fax

Facsimile Modem (T.38 Mode)

coding profile T.38 high speed fax rate

coding profile T.38 low speed packet redundancy

coding profile T.38 high speed packet redundancy

coding profile T.38 data handling method (local/over the network)

Silence Detection

coding profile voice/fax silence detection time

coding profile voice/fax silence signal level

G.711 (A-law/Mu-law)

G.711 coding profile modem coding resampling

Caller ID

caller ID coding profile (on/off) (*Command must set to apply all other caller ID settings*).

voice coding profile

Command Usage

Create coding profile with specified name.

Syntax Options

voice coding profile <*"codingProfName"*>

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦

See Remarks below and details of default coding This command should only be used under the supervision of trained personnel.

Defaults:

For g.729ab, the default profile name is **cp1**.
For g.711 Mu Law, the default profile name is **cp2**.
For fax t.38, the default profile name is **cp3**.
For g.711 A Law, the default profile name is **cp4**.
For g.723.1 63, the default profile name is **cp5**.

Command Example:

voice coding profile salemprof1
voice coding profile calabprof2
voice coding profile cp1
voice coding profile cp2
voice coding profile cp3
voice coding profile cp4
voice coding profile cp5

Remarks

Only one coding profile per codec is allowed. At least one coding profile must be created before activating the voice switching daughtercard. The maximum number of coding profiles allowed per voice switching daughtercard is 128.

Before creating a coding profile, note that there are already five default coding profiles in the master vsmboot.asc file used with the voice daughtercard, namely: cp1, cp2, cp3, cp4 and cp5. If only the default coding profiles are used, then there is no need to create a coding profile. All five default coding profiles are automatically made available for every single voice channel upon power up of the voice switching daughtercard. The default coding profiles each contain preconfigured CLI commands pertaining to the selected codec type. See **view voice coding profile** command for more details and screen output for the command.

There is no default coding profile for modem codecs.

The cp5 coding profile can be used with the OmniPCX by changing the VPI to 30. See the **voice coding profile voice packet interval** command for details.

voice no coding profile

Command Usage

Delete coding profile with specified name.

Syntax Options

voice no coding profile < *"codingProfName"* **>**

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

Default:
None

Command Example:
voice no coding profile salemprof1
voice no coding profile calabprof2
voice no coding profile cp1
voice no coding profile cp2
voice no coding profile cp3
voice no coding profile cp4
voice no coding profile cp5

view voice coding profile

Command Usage

Display coding profile for voice channels.

Syntax Options

view voice coding profile [*"codingProfName"* **]**

Definitions:

codingProfName — Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

Default:
None

Command Example:
view voice coding profile salemprof1
view voice coding profile calabprof2
view voice coding profile
view voice coding profile cp5

Screen Output

To view the default voice coding profiles, type **view voice coding profile**, and then press **<Enter>**. A screen similar to the following displays (shown on next page).

To view a coding profile by another name, type **view coding profile** and a valid coding profile name, e.g., **view voice coding profile calabprof2**, and then press **<Enter>**.

For details on editing the contents of a voice coding profile, default or otherwise, see Chapter 5, "Setup and Installation."

```
****************************************
default voice coding profile
****************************************
****************************************
Voice Coding Profile cp1
****************************************
!
voice coding profile "cp1"
voice coding profile "cp1" codec type g.729ab
voice coding profile "cp1" voice packet interval 30
voice coding profile "cp1" voice activity detector on
voice coding profile "cp1" voice network delay buffer nominal delay 60
voice coding profile "cp1" voice network delay buffer max delay 120
voice coding profile "cp1" voice echo canceller on
voice coding profile "cp1" voice echo canceller non linear on
voice coding profile "cp1" voice echo canceller tail 16
voice coding profile "cp1" voice network delay buffer mode adaptive
voice coding profile "cp1" voice dtmf relay on
voice coding profile "cp1" fax rate 14400
voice coding profile "cp1" call progress tone detection on
voice coding profile "cp1" v.18 tone detection off
voice coding profile "cp1" single frequency tone detection on
voice coding profile "cp1" voice activity detection threshold mode adaptive
voice coding profile "cp1" voice echo canceller comfort noise mode static
voice coding profile "cp1" voice comfort noise level -40
voice coding profile "cp1" echo canceller refresh configuration refresh
voice coding profile "cp1" echo canceller refresh state on
voice coding profile "cp1" caller id off
voice coding profile "cp1" switchover off

!
****************************************
Voice Coding Profile cp2
****************************************
!
voice coding profile "cp2"
voice coding profile "cp2" codec type g.711 mulaw
voice coding profile "cp2" voice packet interval 20
voice coding profile "cp2" voice activity detector on
voice coding profile "cp2" voice network delay buffer nominal delay 80
voice coding profile "cp2" voice network delay buffer max delay 160
voice coding profile "cp2" voice echo canceller on
voice coding profile "cp2" voice echo canceller non linear on
voice coding profile "cp2" voice echo canceller tail 16
voice coding profile "cp2" voice network delay buffer mode adaptive
voice coding profile "cp2" voice dtmf relay on
voice coding profile "cp2" fax rate 14400
voice coding profile "cp2" call progress tone detection on
voice coding profile "cp2" v.18 tone detection off
voice coding profile "cp2" single frequency tone detection on
voice coding profile "cp2" voice activity detection threshold mode adaptive
voice coding profile "cp2" voice echo canceller comfort noise mode static
voice coding profile "cp2" voice comfort noise level -40
voice coding profile "cp2" echo canceller refresh configuration refresh
voice coding profile "cp2" echo canceller refresh state on
voice coding profile "cp2" caller id off
voice coding profile "cp2" switchover off
!
```

```
*****************************************
Voice Coding Profile cp3
*****************************************
!
voice coding profile "cp3"
voice coding profile "cp3" codec type fax t.38
voice coding profile "cp3" fax rate 14400
voice coding profile "cp3" fax transmit level -7
voice coding profile "cp3" fax carrier detect threshold high
voice coding profile "cp3" fax timeout 10
voice coding profile "cp3" fax t.38 high speed packet rate 20
voice coding profile "cp3" fax t.38 low speed redundancy 4
voice coding profile "cp3" fax t.38 high speed redundancy 2
voice coding profile "cp3" fax t.38 training check field method network
voice coding profile "cp3" silence detect level -45
!
*****************************************
Voice Coding Profile cp4
*****************************************
!
voice coding profile "cp4"
voice coding profile "cp4" codec type g.711 alaw
voice coding profile "cp4" voice packet interval 20
voice coding profile "cp4" voice activity detector on
voice coding profile "cp4" voice network delay buffer nominal delay 80
voice coding profile "cp4" voice network delay buffer max delay 160
voice coding profile "cp4" voice echo canceller on
voice coding profile "cp4" voice echo canceller non linear on
voice coding profile "cp4" voice echo canceller tail 16
voice coding profile "cp4" voice network delay buffer mode adaptive
voice coding profile "cp4" voice dtmf relay on
voice coding profile "cp4" fax rate 14400
voice coding profile "cp4" call progress tone detection on
voice coding profile "cp4" v.18 tone detection off
voice coding profile "cp4" single frequency tone detection on
voice coding profile "cp4" voice activity detection threshold mode adaptive
voice coding profile "cp4" voice echo canceller comfort noise mode static
voice coding profile "cp4" voice comfort noise level -40
voice coding profile "cp4" echo canceller refresh configuration refresh
voice coding profile "cp4" echo canceller refresh state on
voice coding profile "cp4" caller id off
!
```

```
*****************************************
Voice Coding Profile cp5
*****************************************
!
voice coding profile "cp5"
voice coding profile "cp5" codec type g.723.1 63
voice coding profile "cp5" voice packet interval 60
voice coding profile "cp5" voice activity detector on
voice coding profile "cp5" voice network delay buffer nominal delay 120
voice coding profile "cp5" voice network delay buffer max delay 240
voice coding profile "cp5" voice echo canceller on
voice coding profile "cp5" voice echo canceller non linear on
voice coding profile "cp5" voice echo canceller tail 16
voice coding profile "cp5" voice network delay buffer mode adaptive
voice coding profile "cp5" voice dtmf relay on
voice coding profile "cp5" fax rate 14400
voice coding profile "cp5" call progress tone detection on
voice coding profile "cp5" v.18 tone detection off
voice coding profile "cp5" single frequency tone detection on
voice coding profile "cp5" voice activity detection threshold mode adaptive
voice coding profile "cp5" voice echo canceller comfort noise mode static
voice coding profile "cp5" voice comfort noise level -40
voice coding profile "cp5" echo canceller refresh configuration refresh
voice coding profile "cp5" echo canceller refresh state on
voice coding profile "cp5" caller id off
voice coding profile "cp5" switchover off
!
```

voice coding profile all reset

Command Usage

Reset *all* coding profiles for voice switching daughtercard to defaults, and delete *all* existing coding profiles at the same time. (*Not available this release.*)

Syntax Options

voice coding profile all reset

Remarks

Coding profile factory default settings are currently available only from the source code.

voice channel available coding profile

Command Usage

Relate coding profile to specified voice channel.

Syntax Options

voice channel <*slot/port/startChannel-endChannel*> **[un]available coding profile**
{ *"codingProfName"* **| all}**

Definitions:

slot	Specifies the chassis slot number where VSM is installed, (e.g., **2**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).
	♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
all	Indicates channel uses all available coding profiles. (*Not available this release.*)

Default:
None

Command Example:
voice channel 2/1/1-12 unavailable coding profile salemprof1
voice channel 2/2/13-24 unavailable coding profile calabprof2
voice channel 2/3/1-30 available coding profile cp1
voice channel 2/4 /1-12 available coding profile calabprof2
voice channel 3/1/13-24 unavailable coding profile all
voice channel 3/2/1-30 available coding profile cp2

Remarks

When **"all"** is used in the syntax of this command, it means that *all* coding profiles are marked accordingly as either available or unavailable for the specified channels.

By default, cp1 through cp5 default coding profiles are automatically made available to all VoIP channels in all ports and slots.

voice channel assign preferred coding profile

Command Usage

Specify preferred coding profile (voice, fax, modem, data) for calls on voice channel.

Syntax Options

voice channel <*slot/port/startChannel-endChannel***> assign preferred {voice | fax | modem} coding profile <** *"codingProfName"* **>**

Definitions:

slot	Specifies the chassis slot number where VSM is installed, (e.g., **2**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**). ♦ **Syntax Note** ♦ Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).
voice	Assigns preferred voice coding profile to channel.
fax	Assigns preferred fax coding profile to channel.
modem	Assigns preferred modem coding profile to channel.
data	Assigns preferred data coding profile to channel.
codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ S % ^ _ & | / \ < > ( ) [ ] { }
all	Indicates channel uses all preferred coding profiles.

Default:
None (see remarks below)

Command Example:
voice channel 2/1/1-12 assign preferred voice coding profile salemprof1
voice channel 2/2/13-24 assign preferred fax coding profile calabprof1
voice channel 2/3/1-30 assign preferred modem coding profile salemprof2

Remarks

Preferred coding profiles must be assigned before activating the voice switching daughtercard.

It is highly recommended that if voice calls are to be processed that this command be issued as it speeds up the process during the h.323 call setup procedure.

By default, cp5 default coding profile is automatically assigned as the preferred voice coding profile for every voice channel on the switch. (See **create coding profile** command.)

By default, cp3 default coding profile is automatically assigned as the preferred fax coding profile for every voice channel on the switch. (See **create coding profile** command.)

There is no default coding profile automatically assigned as the preferred modem coding profile; g.711 for every voice channel on the switch. (See **create coding profile** command.)

voice coding profile coding type

Command Usage

Specify codec type for coding profile. (See table of codec types on next page for more details.)

Syntax Options

voice coding profile < *"codingProfName"* **> codec type <** *codec_type* **>**

Definitions:

codingProfName — Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

codec_type — The specified codec type. Command choices include:

- **g.711 mulaw** (specifies PCM Mu Law audio)
- **g.711 alaw** (specifies PCM A Law audio)
- **g.723.1 53** (specifies g.723.1 5.3 Kbps speech and audio)
- **g.723.1 63** (specifies g.723.1 6.3 Kbps speech and audio)
- **g.729ab** (specifies g.729ab speech and audio (CS-CELP))
- **fax** (specifies fax codec type) *Not available this release*
- **fax t.38** (specifies fax t.38 (real time) codec type)
- **g.726 16** (specifies g.726 16 ADPCM audio) *Not available this release*
- **g.726 24** (specifies g.726 24 Kbps ADPCM audio) *Not available this release*
- **g.726 40** (specifies g.726 40 Kbps ADPCM audio) *Not available this release*
- **g.727 16** (specifies g.727 16 Kbps ADPCM audio) *Not available this release*
- **g.727 24** (specifies g.727 24 Kbps ADPCM audio) *Not available this release*
- **g.727 32** (specifies g.727 32 Kbps ADPCM audio) *Not available this release*
- **g.727 40** (specifies g.727 40 Kbps ADPCM audio) *Not available this release*

Default:
The default codec type is **g.723.1 63** (also referred to as **cp5,** or default coding profile 5).

Command Example:
voice coding profile salemprof1 codec type pcm mulaw
voice coding profile calabprof1 codec type pcm alaw
voice coding profile salemprof2 codec type g.723.1 53
voice coding profile calabprof2 codec type fax t.38
voice coding profile salemprof3 codec type g.726 40
voice coding profile calabprof3 codec type g.727 32

Remarks

PCM stands for Pulse Code Modulation.

ADPCM stands for Adaptive Differential Pulse Code Modulation.

CS-CELP stands for Conjugate Structure Algebraic Code Excited Linear Prediction.

The h.245 (h.323) control functions allow the voice switching daughtercard to negotiate the codec type at runtime.

Use the table below to determine the VPI and VIF for the codec type selected as per the **voice coding profile codec type** command.

Codec Type	Description	Voice	Fax	Modem	Voice Packet Interval (VPI) Time Allowed	Voice Information Field (VIF) Size Reference
g.711 mulaw	G.711 PCM Mu Law	Yes	Yes	Yes	20	640 bits
					30	1280 bits
					40	1920 bits
g.711 alaw	G.711 PCM A Law	Yes	Yes	Yes	20	640 bits
					30	1280 bits
					40	1920 bits
g.723 53 **g.723 63**	G.723.1, 5.3 kbps coding G.723.1, 6.3 kbps coding	Yes	No	No	30 ms	192 bits
					60 ms	384 bits
g.729ab	G.729, Annex A, Annex B 8 kbps coding	Yes	No	No	10 ms	80 bits
					20 ms	160 bits
					30 ms	240 bits
					40 ms	320 bits
					50 ms	400 bits
					60 ms	480 bits
					70 ms	560 bits
					80 ms	640 bits
fax T.38	Fax Relay in T.38 mode	No	Yes	No	not applicable	not applicable

voice coding profile voice packet interval

Command Usage

Specify preferred coding profile voice packet interval (VPI) size and voice information field (VIF) size.

Syntax Options

voice coding profile < *"codingProfName"*> voice packet interval <*packet_size*>

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
packet_size	Specifies voice packet interval in 10 millisecond increments. Supported values include 10, 20, 30, 40, 50, 60, 70, and 80.

Default:
The default packet size is **60**.

Command Example:
voice coding profile salemprof1 voice packet interval 10
voice coding profile calabprof2 voice packet interval 60

Remarks

This command is used to set the size of the voice information field (VIF), in bits, for a coding profile. The VIF size is derived from the specified voice packet interval and the voice coding algorithm. The voice coding algorithm must be specified before the desired voice packet interval.

Each codec type uses a different mathematical algorithm (method) to encode/decode voice (audio) data into IP packets. Any sound that is not a fax, modem, data or signaling tone frequency, including music, is considered voice.

The data or PCM stream coming into the voice channel is referred to as the voice packet interval (VPI) time of the packets coming into the voice channel. The encoder samples the voice traffic every 10 milliseconds, i.e., it determines how many milliseconds of a sample to take, e.g., 10 = 100 ms.

The codec puts the voice data into packet data form using this formula: 10 ms of data is converted into one data packet. The size of the converted packet is the voice information field (VFI) size, and what is ultimately sent over the IP network. The greater the VPI time the larger the VIF packet and the quantity of data transmitted onto the network; however, when the VPI time is shorter, more VIF packets are sent. In other words, the larger the VIF packets the better the quality, but the greater the traffic and use of bandwidth.

voice coding profile voice network delay buffer mode

Command Usage

Set coding profile buffer mode (adaptive/static).

Syntax Options

voice coding profile <*"codingProfName"*> voice network delay buffer mode {adaptive | static}

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
adaptive	Sets voice network buffer delay buffer mode for to adaptive for the specified coding profile.
static	Sets voice network buffer delay mode to static for the specified coding profile.

Default:
The default setting is **static.**

Command Example:
voice coding profile salemprof1 voice network delay buffer mode adaptive
voice coding profile calabprof2 voice network delay buffer mode static

Remarks

This command is used to configure the adaptive playback function mode for coding profiles.

When the voice network delay buffer mode is set to static (adaptive playout disabled), the nominal and maximum playout values are valid.

When the voice network buffer mode is set to adaptive (adaptive playout enabled), the nominal and maximum playout values remain constant, and the DSP adjusts the nominal delay (playout point) to reflect any observed jitter.

The formula for calculating nominal delay (ND) and maximum delay (MD) for the voice playout buffer, given the specified voice packet interval (VPI) size is as follows:

ND / PT = k and MD / PT = j

where:

k and j are integers

k is 2

j is k+2

voice coding profile voice network delay buffer nominal delay

Command Usage

Specify coding profile nominal delay.

Syntax Options

voice coding profile < *"codingProfName"* **> voice network delay buffer nominal delay <***value* **>**

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & ! / \ < > ( ) [ ] { }
value	Specifies the voice network buffer nominal delay, in milliseconds from 1 to 1000, (e.g., **30**).

♦ **Syntax Notes** ♦

Do not use commas when entering the voice network buffer nominal delay value, (for example, **1,000** will return a syntax error message).

The nominal delay should be at least twice the packet interval (in milliseconds); NomDelay = k * packet time, where k > = 2.

Default:
The default value is **120** milliseconds.

Command Example:
voice coding profile salemprof1 voice network buffer delay nominal delay 120
voice coding profile salemprof2 voice network buffer delay nominal delay 500
voice coding profile calabprof1 voice network buffer delay nominal delay 750
voice coding profile calabprof2 voice network buffer delay nominal delay 1000

voice coding profile voice network delay buffer max delay

Command Usage

Specify coding profile maximum delay.

Syntax Options

voice coding profile < *"codingProfName"* **> voice network delay buffer max[imum] delay <** *value* **>**

Definitions:

codingProfName — Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

imum — Optional command syntax. You can type either **max** or **maximum** in the command line.

value — Specifies the voice network buffer maximum delay, in milliseconds from 1 to 1000, (e.g., **30**). The maximum delay should be at least two times greater than the nominal delay (in milliseconds); MaxDelay = k * packet time, where k > j = 2.

♦ **Syntax Notes** ♦

Do not use commas when entering the voice network buffer maximum delay value, (for example, **1,000** will return a syntax error message).

Default:
The default value is **240** milliseconds.

Command Example:
voice coding profile salemprof1 voice network delay buffer maximum delay 240
voice coding profile salemprof2 voice network delay buffer max delay 500
voice coding profile calabprof1 voice network delay buffer max delay 1000

Remarks

Maximum delays must not be greater than the values shown in the following table. (G.726 and G.727 codec types *not available this release.*)

Coding Profile Voice Network Buffer	
Codec	**Maximum Delay**
G.711 64 kbps	160 ms
G.729ab	500 ms
G.723.1	500 ms
G.726, G.727 16 kbps	500 ms
G.726, G.727 24 kbps	370 ms
G.726 G.727 32 kbps	290 ms
G.726, G.727 40 kbps	240 ms

voice coding profile voice activity detector

Command Usage

Set coding profile for voice activity detector (on/off).

Syntax Options

voice coding profile < *"codingProfName"* **> voice activity detector {on | off}**

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
on	Turns ON voice activity detector for specified coding profile.
off	Turns OFF voice activity detector for specified coding profile.

Default:
The default setting is **on**.

Command Example:
voice coding profile salemprof1 voice activity detector on
voice coding profile calabprof2 voice activity detector off

voice coding profile voice activity detection threshold mode

Command Usage

Set coding profile voice activity threshold mode (adaptive/relative).

Syntax Options

voice coding profile < *"codingProfName"* > **voice activity detection threshold mode {adaptive | relative}**

Definitions:

codingProfName — Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

adaptive — Indicates adaptive voice activity detection audio threshold mode for specified coding profile.

relative — Indicates relative voice activity detection audio threshold mode for specified coding profile.

♦ **Syntax Notes** ♦

To use this command, the voice activity detector mode must be enabled via the **voice coding profile voice activity detector** command.

If this command is adaptive in the MPM, the threshold value is automatically 32767 dBm.

If this command is relative in the MPM, the voice activity detection has no bearing on the hardware or VoIP.

Default:
The default setting is **adaptive.**

Command Example:
voice coding profile salemprof1 voice activity detection threshold mode adaptive
voice coding profile calabprof2 voice activity detection threshold mode relative

Remarks

This command is used to set the audio threshold level (in dBm) for the voice activity detector (VAD) for a coding profile to be adaptive or relative to a reference level of -30 dBm.

voice coding profile voice activity detection threshold level

Command Usage

Specify coding profile VAD audio threshold level (adaptive/relative; adaptive if threshold mode enabled).

Syntax Options

voice coding profile < *"codingProfName"* **> voice activity detection threshold level <** *threshold_level* **>**

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & I / \ < > ( ) [ ] { }
threshold_value	Specifies voice activity detection threshold level for specified coding profile.in dBm (decibels below 1 milliwatt; output signal power referenced to 1 milliwatt input signal power). Values may range from -20 through 20 (e.g., **-13**, **-2**, **0**, **4**, **13**, etc).

♦ **Syntax Notes** ♦

To use this command, the voice activity detector mode must be enabled via the **voice coding profile voice activity detector** command.

This command is only valid when the VAD mode is set to relative; if the VAD mode is adaptive, then this command is ignored.

If the supervisory disconnect connection command is turned OFF in the MPM, this value is automatically 32767 dBm.

Default:

The default threshold level is **-13**.

Command Example:

voice coding profile salemprof1 voice activity detection threshold level -13
voice coding profile salemprof2 voice activity detection threshold level 0
voice coding profile calabprof1 voice activity detection threshold level 20
voice coding profile calabprof2 voice activity detection threshold level 1

Remarks

This command is used to set the audio threshold level (in dBm) for the voice activity detector (VAD) for a coding profile to be adaptive or relative to a reference level of -30 dBm.

voice coding profile voice dtmf relay

Command Usage

Set voice coding profile for Dual Tone Multi-Frequency (DTMF) relay (on/off).

Syntax Options

voice coding profile <*"codingProfName"*> voice dtmf relay {on | off}

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
on	Turns ON voice DTMF for specified coding profile.
off	Turns OFF voice DTMF for specified coding profile.

♦ Syntax Notes ♦

This command is only valid with RTP (Realtime Transport Protocol) encapsulation.

Default:
The default setting is **on.**

Command Example:
voice coding profile salemprof1 voice dtmf relay on
voice coding profile calabprof2 voice dtmf relay off

Remarks

DTMF tones are detected during voice processing and separately packetized for transmission.

voice coding profile switchover

Command Usage

Set coding profile fax modem switchover (enable/disable).

Syntax Options

voice coding profile <*"codingProfName"*> switchover {on | off}

Definitions:

codingProfName — Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

on — Turns ON fax/modem switchover for specified coding profile.

off — Turns OFF fax/modem switchover for specified coding profile.

♦ Syntax Notes ♦

If the coding type is set to either PCM Mu Law, PCM A Law, G.726.40 or G.726.40 via the **voice coding profile codec type** command, this parameter can be ON or OFF.

If the coding type is set to fax or fax T.38 this command should be set to OFF, because no switchover is required.

Default:

The default setting is **off**.

Command Example:

voice coding profile salemprof1 switchover off
voice coding profile calabprof2 switchover on

Remarks

DSP tone detection on the voice channel must be ON in the specified coding profile if fax modem switchover is desired, as switchover relies on tone detection. Switchover can be either ON or OFF for a fax-only coding profile.

voice coding profile call progress tone detection

Command Usage

Set coding profile call progress tone detection (on/off).

Syntax Options

voice coding profile <*"codingProfName"*> call progress tone detection {on | off}

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
on	Turns ON secondary level of control over call progress tone detection for specified coding profile.
off	Turns OFF secondary level of control over call progress tone detection for specified coding profile.

♦ **Syntax Notes** ♦

If the call progress detection control for a channel is set to relative via the **call progress tone detection** command, then this parameter determines whether or not detection is enabled.

Default:
The default setting is **off**.

Command Example:
voice coding profile salemprof1 call progress tone detection off
voice coding profile calabprof2 call progress tone detection on

voice coding profile voice dtmf relay

Command Usage

Set voice coding profile for V.18 Annex A call progress tone detection (on/off).

Syntax Options

voice coding profile < *"codingProfName"* **> voice dtmf relay {on | off}**

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ` ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
on	Turns ON V.18 Annex A call progress tone detection for specified coding profile.
off	Turns OFF V.18 Annex A call progress tone detection for specified coding profile.

Default:
The default setting is **off**.

Command Example:
voice coding profile salemprof1 voice dtmf relay off
voice coding profile calabprof2 voice dtmf relay on

Remarks

V.18 Annex A is a 1400 hz. tone used for channel configuration that is detected for 100 ms.

For more information on V.18 Annex A refer to the V.18 Annex A threshold commands.

voice coding profile single frequency tone detection

Command Usage

Set coding profile single frequency call progress tone detection (on/off).

Syntax Options

voice coding profile < *"codingProfName"* **> single frequency tone detection {on | off}**

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
on	Turns ON single frequency call progress tone detection control.
off	Turns OFF single frequency call progress tone detection control.

Default:
The default setting is **off**.

Command Examples:
voice coding profile salemprof1 single frequency tome detection off
voice coding profile calapprof2 single frequency tome detection off

Remarks

The DSPs on the voice switching daughtercard support 2600 hz. tone detection.

voice coding profile voice echo canceller

Command Usage

Set coding profile voice echo canceller (on/off).

Syntax Options

voice coding profile <*"codingProfName"*> voice echo canceller {on | off}

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
on	Turns ON voice echo canceller mode for specified coding profile.
off	Turns OFF voice echo canceller mode for specified coding profile.

Default:
The default setting is **on**.

Command Example:
voice coding profile salemprof1 voice echo canceller on
voice coding profile calapprof2 voice echo canceller off

Remarks

Voice switching daughtercards perform echo removal on PCM samples using a proprietary double filter algorithm that provides stability and performance up to 128 ms echo cancellation tail length. See also ITU-T Recommendation G.165: Echo Cancellers.

For more information on echo cancellers, refer to the echo and acoustic echo cancellation commands used in Telephony Signaling templates.

voice coding profile voice echo canceller non linear

Command Usage

Set coding profile voice echo canceller non-linear processor mode (on/off).

Syntax Options

voice coding profile <*"codingProfName"*> voice echo canceller non linear {on | off}

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
on	Turns ON non-linear voice echo canceller processor mode for specified coding profile.
off	Turns OFF non-linear voice echo canceller processor mode for specified coding profile.

♦ **Syntax Notes** ♦

To use this command, the voice echo canceller mode must be enabled via the **coding profile voice echo canceller** command.

Default:
The default setting is **on**.

Command Example:
voice coding profile salemprof1 voice echo canceller non linear on
voice coding profile calabprof2 voice echo canceller non linear off

voice coding profile voice echo canceller comfort noise mode

Command Usage

Specify coding profile voice echo canceller comfort noise mode.

Syntax Options

voice coding profile <*"codingProfName"*> voice echo canceller comfort noise mode {static | adaptive}

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
static	Turns ON the fixed voice comfort noise level.
adaptive	Turns OFF the fixed voice comfort noise level at runtime for the duration of the call, if it is determined that the phone is digital (optional selection).

♦ Syntax Notes ♦

To use this command, the voice echo canceller mode must be enabled via the **coding profile voice echo canceller** command.

Default:
The default setting is **static**.

Command Example:
voice coding profile salemprof1 voice echo canceller comfort noise mode static
voice coding profile calabprof2 voice echo canceller comfort noise mode adaptive

voice coding profile voice echo canceller noise level

Command Usage

Specify coding profile voice echo canceller noise level.

Syntax Options

voice coding profile < *"codingProfName"* **> voice echo canceller noise level <***value* **>**

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
value	Specifies voice echo canceller noise level for specified coding profile, in dBm, (power referenced to 1 milliwatt input signal power), from **-70** to **-40**, (e.g., **-40**).

♦ Syntax Notes ♦

To use this command, the voice echo canceller mode must be enabled via the **coding profile voice echo canceller** command. The voice echo canceller comfort noise mode must also be set to static via the **coding profile voice echo canceller comfort noise mode** command.

Default:

The default value is **-40**.

Command Example:

voice coding profile salemprof1 voice echo canceller noise level -40
voice coding profile calabprof2 voice echo canceller noise level -70

voice coding profile voice echo canceller tail length

Command Usage

Specify coding profile voice echo canceller tail delay length.

Syntax Options

voice coding profile < *"codingProfName"* **> voice echo canceller tail length <** *value* **>**

Definitions:

codingProfName — Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

value — Specifies voice echo canceller tail length for specified coding profile, in milliseconds, from **0** to **16,** (e.g., **16**); **4, 8, 16** are the only legal values for this release.

♦ **Syntax Notes** ♦

To use this command, the voice echo canceller mode must be enabled via the **coding profile voice echo canceller** command.

If the voice echo canceller is OFF in the MPM, then a **0** tail delay length is automatically generated.

Default:
The default value is **16.**

Command Example:
voice coding profile salemprof1 voice echo canceller tail length 16
voice coding profile calabprof2 voice echo canceller tail length 0
voice coding profile salemprof1 voice echo canceller tail length 128

Remarks

voice coding profile voice echo canceller refresh configuration

Command Usage

Specify coding profile voice echo canceller refresh configuration state.

Syntax Options

voice coding profile < *"codingProfName"* > voice echo canceller refresh configuration {frozen | refresh}

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
frozen	Specifies voice echo canceller configuration refresh state for specified coding profile.
refresh	Specifies voice echo canceller configuration refresh state is frozen for specified coding profile.

Default:
The default setting is **frozen.**

Command Example:
voice coding profile salemprof1 voice echo canceller refresh configuration frozen
voice coding profile calabprof2 voice echo canceller refresh configuration refresh

Remarks

In the configuration refresh state the echo canceller automatically adjusts as the call progresses based upon the PCM sample.

voice coding profile voice echo canceller refresh state

Command Usage

Specify coding profile voice echo refresh state.

Syntax Options

voice coding profile < *"codingProfName"* **> voice echo canceller refresh state {on | off}**

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
on	Turns ON (resets) voice echo canceller refresh state for specified coding profile.
off	Turns OFF voice echo canceller refresh state for specified coding profile (normal state).

Default:
The default setting is **on**.

Command Example:
voice coding profile salemprof1 voice echo canceller refresh state on
voice coding profile calabprof2 voice echo canceller refresh state off

voice coding profile fax rate

Command Usage

Specify coding profile maximum allowed fax modem data rate.

Syntax Options

voice coding profile < *"codingProfName"* **> fax rate {2400 | 4800 | 7200 | 9600 | 12000 | 14400}**

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
2400	Specifies a maximum fax data baud rate of 2400 bps.
4800	Specifies a maximum fax data baud rate of 4800 bps.
7200	Specifies a maximum fax data baud rate of 7200 bps.
9600	Specifies a maximum fax data baud rate of 9600 bps.
12000	Specifies a maximum fax data baud rate of 12000 bps.
14400	Specifies a maximum fax data baud rate of 14400 bps.

Default:
The default baud rate is **14400**.

Command Example:
voice coding profile salemprof1 fax rate 14400
voice coding profile calabprof2 fax rate 9600

voice coding profile fax transmit level

Command Usage

Specify coding profile fax modem transmit level gain.

Syntax Options

voice coding profile < *"codingProfName"*> fax transmit level <*gain_value*>

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & ǀ / \ < > ( ) [ ] { }
gain_value	Specifies fax modem transmit (TX) gain for specified voice coding profile, in decibels. Values may range from -13 through 0 (e.g., **-13**, **-2**, etc).

Default:
The default value is **-13** decibels.

Command Example:
voice coding profile salemprof1 fax transmit level -13
voice coding profile calabprof2 fax transmit level 0

voice coding profile fax carrier detect threshold

Command Usage

Specify coding profile fax modem carrier detect threshold.

Syntax Options

voice coding profile <*"codingProfName"*> fax carrier detect threshold {low | medium | high}

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ &	/ \ < > ( ) [ ] { }
low	Sets fax modem carrier detect threshold to -43 dBm for specified coding profile.
medium	Sets fax modem carrier detect threshold to -33 dBm for specified coding profile.
high	Sets fax modem carrier detect threshold to -26 dBm for specified coding profile.

Default:
The default setting is **high**.

Command Example:
voice coding profile salemprof1 fax carrier detect threshold high
voice coding profile salemprof2 fax carrier detect threshold medium
voice coding profile calabprof1 fax carrier detect threshold low

voice coding profile fax timeout

Command Usage

Specify coding profile inactivity detection time to automatically tear down fax.

Syntax Options

voice coding profile < *"codingProfName"* **> fax timeout <***value* **>**

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
value	Specifies voice coding profile fax timeout (no activity time) on fax modem connection before call is cleared, in milliseconds, from 10 to 32,000 seconds, (e.g., **20**).

♦ Syntax Note ♦

Do not use commas when entering the voice coding profile fax timeout value, (for example, **1,000** will return a syntax error message).

Default:
The default value is **20** milliseconds.

Command Example:
voice coding profile salemprof1 fax time out value 20
voice coding profile salemprof2 fax time out value 12000
voice coding profile calabprof2 fax time out value 32000

voice coding profile fax t.38 high speed packet rate

Command Usage

Specify coding profile T.38 high speed fax rate.

Syntax Options

voice coding profile <*"codingProfName"*> **fax t.38 high speed packet rate {10 | 20 | 30 | 40}**

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
10 | 20 | 30 | 40	Specifies rate at which high speed data is sent across network for specified fax coding profile.

Default:
The default fax rate is **20**.

Command Example:
voice coding profile salemprof1 fax t.38 high speed packet rate 20
voice coding profile calabprof2 fax t.38 high speed packet rate 40

voice coding profile fax t.38 low speed redundancy

Command Usage

Specify coding profile T.38 low speed packet redundancy.

Syntax Options

voice coding profile <*“codingProfName”*> fax t.38 low speed redundancy {0 | 1 | 3 | 4 | 5}

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
0 | 1 | 3 | 4 | 5	Specifies packet-level redundancy for low speed data transmission (i.e., T.30 handshaking info) for specified fax coding profile.

Default:
The default value is **4**.

Command Example:
voice coding profile salemprof1 fax t.38 low speed redundancy 4
voice coding profile calabprof2 fax t.38 low speed redundancy 5

voice coding profile fax t.38 high speed redundancy

Command Usage

Specify coding T.38 high speed packet redundancy.

Syntax Options

voice coding profile < *"codingProfName"* **> fax t.38 high speed redundancy {0 | 1 | 2}**

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
0 | 1 | 2	Specifies packet-level redundancy for high speed data transmission (i.e., T.4 image data) for specified fax coding profile.

Default:
The default value is **2**.

Command Example:
voice coding profile salemprof1 fax t.38 high speed redundancy 1
voice coding profile calabprof2 fax t.38 high speed redundancy 2

voice coding profile fax t.38 training check field method

Command Usage

Specify coding profile T.38 data handling method (local/over the network).

Syntax Options

voice coding profile < *"codingProfName"* > fax t.38 training check field method {local | network}

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
local	Specifies local method of handling data over the network for specified coding profile.
network	Turns network method of handling data over the network for specified coding profile.

Default:
The default setting is **network.**

Command Example:
voice coding profile salemprof1 fax t.38 training check field method local
voice coding profile calabprof2 fax t.38 training check field method network

Remarks

The local method (method 1) requires that the training check field (TCF) training signal be generated and checked locally by the gateway, and not be forwarded over the network. With the network method (method 2), TCF data is sent over the network. Both methods correspond to data management methods 1 and 2 in the T.38 UDP fax protocol specification.

voice coding profile silence detect time

Command Usage

Specify voice/fax coding profile silence detection time.

Syntax Options

voice coding profile < *"codingProfName"* > [no] silence detect time <*value* >

Definitions:

codingProfName — Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and
* ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

value — Specifies voice/fax coding profile silence detection time, in milliseconds, from 5 to 32,000, (e.g., **2000**).

♦ **Syntax Notes** ♦

Do not use commas when entering the value for voice/fax coding profile silence detection time, (for example, **2,000** will return a syntax error message).

no — Disables silence detection. A "no silence detect time 100" or any other number of milliseconds disables silence detection.

Default:
The default value is **5** milliseconds.

Command Example:

voice coding profile salemprof1 no silence detect time
voice coding profile salemprof1 silence detect time value 100
voice coding profile calabprof2 silence detect time value 100

voice coding profile silence detect level

Command Usage

Specify voice/fax coding profile silence signal level.

Syntax Options

voice coding profile <"*codingProfName*"**> silence detect level <***signal_level***>**

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ` ' , : ; .@ $ % ^ _ & | / \ < > ( ) [ ] { }
signal_level	Specifies voice/fax coding profile silence signal level, in decibels. Values may range from -50 through -40 (e.g., **-42**, **-50**, **-40**, etc).

Default:
The default value is **-50** decibels.

Command Example:
voice coding profile salemprof1 silence detect level -50
voice coding profile calabprof2 silence detect level -40

voice coding profile g.711 modem resampling mode

Command Usage

Specify g.711 (PCM Mu Law/PCM A Law) modem coding resampling.

Syntax Options

voice coding profile <*"codingProfName"*> g.711 modem resampling mode {on | off}

Definitions:

codingProfName	Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
on	Turns ON g.711 (Mu Law/A Law) modem resampling mode for specified coding profile.
off	Turns OFF g.711 (Mu Law/A Law) modem resampling mode for specified coding profile.

♦ **Syntax Notes** ♦

The following settings must be made to use this command:

The voice coding profile codec type must be set to g.711 via the **voice coding profile codec type** command.

The voice coding profile adaptive playout delay must be disabled via the **voice network delay buffer mode** command.

The voice coding profile maximum network buffer delay must be set to maximum via the **voice network delay buffer maximum delay** command.

The voice coding profile nominal network buffer delay must be set to half of the maximum buffer delay via the **network delay buffer nominal delay** command.

The voice activity detection must be disabled via the **voice coding profile voice activity detector** command.

Default:
The default setting is **off**.

Command Example:
voice coding profile salemprof1 g.711 modem resampling on
voice coding profile calabprof2 g.711 modem resampling off

voice coding profile caller id

Command Usage

Set caller ID for specified coding profile (on/off); this command must be set to apply all other caller ID settings as listed:

<u>Telephony Signaling</u>

FXS LS to generate outbound caller ID (on/off)

FXS LS to detect inbound caller ID (on/off)

FXO GS to generate outbound caller ID (on/off)

FXO GS to detect inbound caller ID (on/off)

<u>Outbound Caller ID</u>

Outbound caller ID name (private/unavailable) to transmit

Outbound caller ID number (published/unpublished) to transmit

Syntax Options

voice coding profile <*"codingProfName"*> caller id {on | off}

<u>Definitions</u>:

codingProfName	Identifies the coding profile by name, (e.g., **cprofcallid1**). Consists of at least one ASCII character with quotes on each end of the name; maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & I / \ < > ( ) [ ] { }
on	Turns ON caller ID for specified coding profile.
off	Turns OFF caller ID for specified coding profile.

<u>Default</u>:
The default setting is **on**.

<u>Command Examples</u>:
voice coding profile "cprofcallid1" caller id off
voice coding profile "cprofcallid2" caller id on
voice coding profile "cprofcallid3" caller id off
voice coding profile "cprofcallid4" caller id on
voice coding profile "cprofcallid5" caller id off
voice coding profile "cprofcallid6" caller id on

Voice Network Template

The commands listed and described below are used to assign and configure the Voice Network Template and the following related components: H.323 gateway discovery, operations and configuration.

Voice Network Template (create, delete and view)

Voice Network Template (assign to voice switching daughtercard) (*Not available this release.*)

H.323 Gateway Discovery

gatekeeper control (on/off)

gatekeeper discovery mode (manual/off) (Auto discovery *not available this release.*)

gatekeeper IP address for gatekeeper discovery (manual mode only)

H.323 Gateway Configuration

calls allowed (or disallowed) without gatekeeper (no gateway endpoint regis.; true/false)

no. of registration attempts allowed (before gateway endpoint registration failure)

gateway endpoint registration type (if gatekeeper used)

associate (or disassociate) phone groups with gatekeeper (if gatekeeper used)

H.323 Gateway Operations

H.323 display name for voice switching daughtercard gateway

RTP/RTCP *port mode* for voice switching daughtercard gateway (dynamic/sequential)

starting RTP/RTCP *port number* for voice switching daughtercard gateway (if sequential)

voice network template

Command Usage

Create voice network template with specified network interface name.

Syntax Options

voice network template < *"TemplateName"* **>**

Definitions

TemplateName — Identifies the voice network template created by name, (e.g., **vsmvon1**); maximum length of 40 characters. The following characters are permitted in the voice network template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }.

Default:
None

Command Example:
voice network template salemvon1
voice network template calabvon2

Remarks

Voice network templates are used to configure the parameters of the call control protocol of the voice traffic over the network, and are assigned individually to voice switching daughtercards. Voice network templates must be created before activating the voice switching daughtercard.

no voice network template

Command Usage

Delete voice network template with specified network interface name.

Syntax Options

no voice network template < *"TemplateName"* >

Definitions:

TemplateName	Identifies the voice network template by name, (e.g., **vsmvon1**); maximum length of 40 characters. The following characters are permitted in the voice network template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & I / \ < > ( ) [ ] { }

Default:
None

Command Example:
voice network template salemvon1
voice network template salemvon2
no voice network template calabvon1
no voice network template calabvon2

Remarks

Voice network templates are used to configure the parameters of the call control protocol used by the voice traffic sent over the network. The templates are assigned individually to voice switching daughtercards.

view voice network template

Command Usage

Display voice network template with specified network interface name.

Syntax Options

view voice network template < *"TemplateName"* >

Definitions:

TemplateName — Identifies the voice network template by name, (e.g., **vsmvon1**); maximum length of 40 characters. The following characters are permitted in the voice network template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

Default:

None

Command Example:

view voice network template salemvon1
view voice network template calabvon2

Remarks

This command displays all configured parameters of the specified voice network template.

Screen Output

To view the parameters of a voice network template, type **view voice network template** followed by a valid voice network template name, e.g., **view voice network template use gateway**, and then press **<Enter>**.

A screen similar to the following displays.

```
*******************************************
Viewing Network Template
*******************************************
!
voice network template use gatekeeper
voice network template use gatekeeper display name gatekeeper
voice network template use gatekeeper h.323 rtp port mode sequential
voice network template use gatekeeper h.323 rtp port base 30060
voice network template use gatekeeper h.323 gatekeeper control on
voice network template use gatekeeper h.323 gatekeeper mode manual
voice network template use gatekeeper h.323 gatekeeper address 195.167.10.33
voice network template use gatekeeper h.323 allow calls without gatekeeper false
voice network template use gatekeeper h.323 gatekeeper maximum tries 24
voice network template use gatekeeper h.323 endpoint registration type gateway
!
```

voice daughter card assign network template

Command Usage

Assign voice network template to specified voice switching daughtercard. (*Not available this release;* all commands using the syntax *"vsmNetworkTemplateName* are currently not applicable as a result).

Syntax Options

voice daughter card <*slot/card_number*> **assign network template** < *"TemplateName"* >

Definitions:

slot	Specifies the chassis slot number where VSM is installed, (e.g., **2**).
card_number	Specifies the voice daughtercard position number, (e.g., **1**).
TemplateName	Identifies the voice network profile by name, (e.g., **vsmvon1**); maximum length of 40 characters. The following characters are permitted in the voice network template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

Default:
None

Command Example:
voice daughter card 2/1 assign network template salemvon1
voice daughter card 2/2 assign network template salemvon2
voice daughter card 2/3 assign network template calabvon1
voice daughter card 2/4 assign network template calabvon2

Remarks

Refer to the voice switching daughtercard activation command for information on how to activate a card once it has been assigned a voice network template, and for details on using the faststart mode gateway commands in relation to voice network templates.

voice network h.323 gatekeeper control

Command Usage

Set voice switching daughtercard to use gatekeeper (on/off).

Syntax Options

voice network {template *"TemplateName"* **| card** *slot/card_number* **} h.323 gatekeeper control {on | off}**

Definitions:

TemplateName	Identifies the voice network template by name, (e.g., **vsmvon1**); maximum length of 40 characters. The following characters are permitted in the voice network template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _& | / \ < > ( ) [ ] { }
slot	Specifies the chassis slot number where VSM is installed, (e.g., **2**).
card_number	Specifies the voice daughtercard position number, (e.g., **1**).
on	Turns ON H.323 gatekeeper for specified voice network template.
off	Turns OFF H.323 gatekeeper for specified voice network template.

Default:
The default setting is **off**.

Command Example:
voice network template salemvon1 h.323 gatekeeper control off
voice network template salemvon2 h.323 gatekeeper control on
voice network card 2/1 h.323 gatekeeper control off
voice network card 2/2 h.323 gatekeeper control on

voice network h.323 gatekeeper mode

Command Usage

Set gatekeeper mode for voice switching daughtercard gateway discovery (manual/auto). (*Not available this release.*)

Syntax Options

voice network {template *"TemplateName"* **| card** *slot/card_number* **} h.323 gatekeeper mode {manual | auto}**

Definitions:

TemplateName	Identifies the voice network template by name, (e.g., **vsmvon1**); maximum length of 40 characters. The following characters are permitted in the voice network template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies the chassis slot number where VSM is installed, (e.g., **2**).
card_number	Specifies the voice daughtercard position number, (e.g., **1**).
manual	Turns ON H.323 gatekeeper manual discovery mode for specified voice network template.
auto	Turns ON H.323 gatekeeper autodiscovery mode for specified voice network template.

♦ **Syntax Note** ♦

To use this command, the **h.323 gatekeeper control** command must be turned ON.

Default:

The default setting is **manual**.

Command Example:

voice network template salemvon1 h.323 gatekeeper mode manual
voice network template salemvon2 h.323 gatekeeper mode auto
voice network card 2/1 h.323 gatekeeper mode manual
voice network card 2/2 h.323 gatekeeper mode auto

Remarks

This command is used to control gatekeeper operations in conjunction with the **h.323 gatekeeper control** command. Internally, the setting affects two different variables in the switch configuration. These include the gatekeeper configuration field which controls the enable/disable operation of the gatekeeper, and the auto discovery gatekeeper configuration field (when enabled), which determines automatic or manual discovery.

voice network h.323 gatekeeper address

Command Usage

Specify gatekeeper IP address for voice switching daughtercard gateway discovery (manual mode only).

Syntax Options

voice network {template *"TemplateName"* **| card** *slot/card_number* **} h.323 gatekeeper address** <*ip_address*>

Definitions:

TemplateName	Identifies the voice network template by name, (e.g., **vsmvon1**); maximum length of 40 characters. The following characters are permitted in the voice network template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }.
slot	Specifies the chassis slot number where VSM is installed, (e.g., **2**).
card_number	Specifies the voice daughtercard position number, (e.g., **1**).
ip_address	Specifies IP address of the voice daughtercard H.323 gatekeeper, (e.g., **224.0.1.41**).

♦ Syntax Notes ♦

H.323 display name string must be specified before the voice switching daughtercard is activated. A non-null string value is required.

To use this command, the **h.323 gatekeeper mode** command for gatekeeper discovery must be set to MANUAL.

Default:
None

Command Example:
voice network template salemvon1 h.323 gatekeeper address 224.0.1.41
voice network template calabvon2 h.323 gatekeeper address 224.0.1.42
voice network card 2/1 h.323 gatekeeper address 224.0.1.41
voice network card 2/2 h.323 gatekeeper address 224.0.1.42

Remarks

This command is used to specify the address of the gatekeeper in the currently active (H.323) zone when configured for manual mode. Because port 1719 is used, only the IP address needs to be specified.

voice network h.323 allow calls without gatekeeper

Command Usage

Set calls allowed (or disallowed) without gatekeeper; voice switching daughtercard gateway endpoint not registered (true/false).

Syntax Options

voice network {template *"TemplateName"* **| card** *slot/card_number* **} h.323 allow calls without gatekeeper {true | false}**

Definitions:

TemplateName	Identifies the voice network template by name, (e.g., **vsmvon1**); maximum length of 40 characters. The following characters are permitted in the voice network template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies the chassis slot number where VSM is installed, (e.g., **2**).
card_number	Specifies the voice daughtercard position number, (e.g., **1**).
true	H.323 allows calls without gatekeeper.
false	H.323 does not allow calls without gatekeeper.

Default:
The default setting is **true**.

Command Example:
voice network template salemvon1 h.323 allow calls without gatekeeper true
voice network template calabvon2 h.323 allow calls without gatekeeper false
voice network card 2/1 h.323 allow calls without gatekeeper true
voice network card 2/2 h.323 allow calls without gatekeeper false

voice network h.323 allow calls without gatekeeper max tries

Command Usage

Specify number of registration attempts allowed before voice switching daughtercard gateway endpoint registration failure occurs.

Syntax Options

voice network {template *"TemplateName"* **| card** *slot/card_number* **} h.323 allow calls without gatekeeper max[imum] tries <***value***>**

Definitions:

TemplateName	Identifies the voice network template by name, (e.g., **vsmvon1**); maximum length of 40 characters. The following characters are permitted in the voice network template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies the chassis slot number where VSM is installed, (e.g., **2**).
card_number	Specifies the voice daughtercard position number, (e.g., **1**).
imum	Optional command syntax. You can type either **max** or **maximum** in the command line.
value	Specifies number of registration attempts made by voice switching daughtercard before it is allowed to fail registration, (e.g., 4).

Default:
The default value is **4**.

Command Example:
voice network template salemvon1 h.323 allow calls without gatekeeper maximum tries 1
voice network template salemvon2 h.323 allow calls without gatekeeper max tries 4
voice network card 2/1 h.323 allows calls without gatekeeper maximum tries 1
voice network card 2/2 h.323 allows calls without gatekeeper max tries 4

Remarks

Once the number of unsuccessful registration attempts is passed, the endpoint is only able to place calls if the **h.323 allow calls without gatekeeper** command is set to **true**.

voice network h.323 endpoint registration type

Command Usage

Specify voice switching daughtercard gateway endpoint registration type (if gatekeeper used).

Syntax Options

voice network {template *"TemplateName"* **| card** *slot/card_number* **} h.323 endpoint registration type {gateway | terminal}**

Definitions:

TemplateName	Identifies the voice network template by name, (e.g., **vsmvon1**); maximum length of 40 characters. The following characters are permitted in the voice network template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies the chassis slot number where VSM is installed, (e.g., **2**).
card_number	Specifies the voice daughtercard position number, (e.g., **1**).
gateway	Enables gateway endpoint registration.
terminal	Enables terminal endpoint registration.

♦ Syntax Note ♦

The only time this command should be set to **terminal** is when an analog voice switching daughtercard is in use, and only one phone number and one analog port are in service.

Default:
The default setting is **gateway**.

Command Example:
voice network template salemvon1 h.323 endpoint registration type gateway
voice network template salemvon2 h.323 endpoint registration type terminal
voice network card 2/1 h.323 endpoint registration type gateway

Remarks

This command is used to set the H.225.0 endpoint registration type of the voice switching daughtercard. This should not be confused with the H.245 terminal type, although the two parameters should be programmed consistently. This parameter specifies how the endpoint will register itself with the gatekeeper, and has nothing to do with master/slave determination. See the RadVision H.323 Gatekeeper User Manual for more information.

voice network h.323 gatekeeper associate

Command Usage

This command is used to associate or (disassociate) one or more phone groups with a daughtercard gatekeeper, thereby enabling the daughtercard to generate the legal h.323 alias names that are sent to the gatekeeper in lieu of a telephone number.

Syntax Options

voice network {template *"TemplateName"* **| card** *slot/card_number* **} h.323 gatekeeper [dis]associate [phone group] <** *"phoneGrpName"* **>**

Definitions:

TemplateName	Identifies the voice network template by name, (e.g., **vsmvon1**); maximum length of 40 characters. The following characters are permitted in the voice network template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies the chassis slot number where VSM is installed, (e.g., **2**).
card_number	Specifies the voice daughtercard position number, (e.g., **1**).
phone group	Optional command syntax.
phoneGrpName	Specifies the phone group to which the gatekeeper is associated (or disassociated).

♦ **Syntax Notes** ♦

This command must be issued if the H.323 **gatekeeper control** command is ON.

In order for the voice daughtercard to generate alias names using this command, either the **voice daughter card activate** command or the **voice numbering plan activate** command must be issued. (The **voice numbering plan activate** command is *not available this release.*)

Default:
None

Command Example:
voice network template vsmvon1 h.323 gatekeeper associate phone group salem_engr1
voice network template vsmvon2 h.323 gatekeeper associate salem_engr1
voice network template vsmvon1 h.323 gatekeeper disassociate phone group salem_engr1
voice network template vsmvon2 h.323 gatekeeper disassociate salem_engr1
voice network card 2/1 h.323 gatekeeper associate phone group salem_engr1
voice network card 2/2 h.323 gatekeeper associate salem_engr1
voice network card 2/1 h.323 gatekeeper disassociate phone group salem_engr1
voice network card 2/2 h.323 gatekeeper disassociate salem_engr1

Remarks

Only E.164 alias names are generated by the gateway. E.164 is an ITU ISDN/SMDS (Switched Multimegabit Data Service) phone line numbering scheme; SDMS is used in LAN to LAN metropolitan networks.

voice network h.323 display name

Command Usage

Specify H.323 display name for voice switching daughtercard gateway.

Syntax Options

voice network {template *"TemplateName"* **| card** *slot/card_number* **} h.323 display name <** *"string"* **>**

Definitions:

TemplateName	Identifies the voice network template by name, (e.g., **vsmvon1**); maximum length of 40 characters. The following characters are permitted in the voice network template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies the chassis slot number where VSM is installed, (e.g., **2**).
card_number	Specifies the voice daughtercard position number, (e.g., **1**).
string	Any string up to 64 bytes, (e.g., **salemgway**). The display name is carried as an H.323_ID alias name.

♦ Syntax Note ♦

H.323 display name string must be specified before the voice switching daughtercard is activated. A non-null string value is required.

Default:
None

Command Example:
voice network template salemvon1 h.323 display name salemgway
voice network template calabvon2 h.323 display name calabgway
voice network card 2/1 h.323 display name salemgway
voice network card 2/2 h.323 display name calabgway

Remarks

This command is used to set the display name information that is carried in the H.323 setup messages. The display name string is inserted into the Q.931 display information and source address field of the H.323 setup (UUIE).

voice network h.323 rtp port mode

Command Usage

Set RTP/RTCP (Real Time Protocol/Real Time Conferencing Protocol) port mode for voice switching daughtercard H.323 gateway (dynamic/sequential).

Syntax Options

voice network {template *"TemplateName"* **| card** *slot/card_number* **} h.323 rtp port mode {dynamic | sequential}**

Definitions:

TemplateName	Identifies the voice network template by name, (e.g., **vsmvon1**); maximum length of 40 characters. The following characters are permitted in the voice network template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
slot	Specifies the chassis slot number where VSM is installed, (e.g., **2**).
card_number	Specifies the voice daughtercard position number, (e.g., **1**).
dynamic	Assigns RTP/RTCP port numbers dynamically for H.323 gateway.
sequential	Assigns RTP/RTCP port numbers sequentially for H.323 gateway.

♦ Syntax Notes ♦

If the starting RTP/RTCP port mode for the gateway is set to dynamic, then the RTP port base value is automatically set to **0**.

Starting RTP/RTCP port numbers are specified via the **h.323 RTP port base** command.

Default:
The default setting is **dynamic**.

Command Example:
voice network template salemvon1 h.323 rtp port mode dynamic
voice network template calabvon2 h.323 rtp port mode sequential
voice network card 2/1 h.323 rtp port mode dynamic
voice network card 2/2 h.323 rtp port mode sequential

Remarks

This command is used to set the port number assignment method for RTP and RTCP ports.

voice network h.323 rtp port base

Command Usage

Specify starting RTP/RTCP port number for voice switching daughtercard gateway (if sequential).

Syntax Options

voice network {template *"TemplateName"* **| card** *slot/card_number* **} h.323 rtp port base <***value***>**

Definitions·

TemplateName — Identifies the voice network template by name, (e.g., **vsmvon1**); maximum length of 40 characters. The following characters are permitted in the voice network template name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

slot — Specifies the chassis slot number where VSM is installed, (e.g., **2**).

card_number — Specifies the voice daughtercard position number, (e.g., **1**).

value — Specifies starting RTP or RTCP port value from 1 to 65535, (e.g., **30000**). If dynamic port assignment is preferred, the RTP port base should be set to a value of **0**; if dynamic port assignment is used, the requirements of the H.323 specification many not be met, as no attempt is made to assure proper numbering upon assignment.

♦ Syntax Notes ♦

Do not use commas when entering the starting value for RTP/RTCP ports, (for example, **30,000** will return a syntax error message).

To use this command, the starting RTP/RTCP port mode for the gateway must be set to sequential via the **RTP port mode** command.

If the starting RTP/RTCP port mode for the gateway is set to dynamic, then this value is ignored.

Default:

The default value is **30000**.

Command Example:

voice network template salemvon1 h.323 rtp port base 30000
voice network template salemvon2 h.323 rtp port base 40000
voice network card 2/1 calabvon1 h.323 rtp port base 50000
voice network card 2/2 calabvon2 h.323 rtp port base 65535

Remarks

This command is used to specify the starting port number assigned to RTP/RTCP ports; when H.323 calls are made an RTP or RTCP port is opened for each call. The RTP port number should be an even number, and the RTCP port number should be one number greater than the RTP port value.

In order to accomplish this sort of controlled allocation, the port numbers are assigned starting at the RTP port base value. Call Control Block (CCB) 0 will use ports numbered RTP port base and RTP port base + 1. CCB 1 will use the next two successive ports, etc. When the call is terminated, the CCB number will eventually be reused.

For each new H.323 call, the CCB number is incremented. The first call would be CCB 0, the second would be CCB 1 and so on, as shown below where the RTCP port number is set one number higher than the RTP port number, e.g., the starting RTP port number is 30001 (with a base of 30000), and the starting RTCP port number is 30002.

- On the first incoming H.323 call setup message, the starting RTP port number would be 30003, and the starting RTCP port number would be 30004.
- On the next incoming H.323 call setup message, the starting RTP port number would be 30004, and the starting RTCP port number would be 30005.
- On the subsequent incoming H.323 call setup message, the starting RTP port number would be 30006, and the starting RTCP port number would be 30007, and so on.

Standard port assignments are as follows:

Gatekeeper User Datagram Protocol (UDP) Discovery Multicast Address: 224.0.1.41

Gatekeeper UDP Discovery Port (automatic discovery): 1718

Gatekeeper UDP Registration and Status Port (manual discovery): 1719

Endpoint Transmission Control Protocol (TCP) Call Signaling Port: 1720

Network Dialing Scheme

The commands listed and described below are used to configure the Network Dialing Scheme and related components as follows: destinations, phone groups and phone group parameters, and numbering plans including numbering plan hunt methods and descriptions.

Destinations

H.323 endpoint destination name

local channel destination

delete destination

view destination

Phone Groups

create (or delete) voice phone group

view voice phone group

Inbound/Outbound Digit Processing

unique phone group site *prefix* for routing VoIP calls (on/off)

unique phone group site *prefix digits* for routing VoIP calls

voice phone group dialing type

voice phone group format of tel. number and number of outbound digits to dial

Additional Inbound/Outbound Digit Processing

number of outbound digits to strip in voice phone group site (before forwarding call)

allow forwarding of phone group prefix

phone group site digits to prefix (before forwarding call)

Call Access Type

type of access calls (voice, fax, modem, data) allowed in phone group (on/off)

Digit Dialing Ranges

voice phone group site *numbers to include* in range of digits for phone format string

voice phone group site *numbers to remove* from range of digits for phone format string

Numbering Plan

create (or delete) numbering plan

view voice numbering plan

activate voice numbering plan

Numbering Plan Hunt Method

outgoing hunting method of voice numbering plan (destinations group)

Numbering Plan Description

voice numbering plan (optional)

Associate Numbering Plan

associate (or disassociate) *destinations* with numbering plans

associate (or disassociate) *phone groups* with numbering plans

voice destination h.323 endpoint

Command Usage

Create an endpoint destination with specified name that uses the H.323 protocol.

Syntax Options

voice destination < *"endpointDestName"* **> h.323 endpoint <** *address* **>** [*port*]

Definitions:

endpointDestName	Identifies the voice call endpoint destination string name unique across the H.323 network, (e.g., **to_vsd1**) any string up to 64 bytes (not including quotes). The following characters are permitted in the destination string name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & I / \ < > ( ) [ ] { }
address	H.323 transport address, i.e., IP or H.323 network address, (e.g., **225.0.1.41**).
port	Any h.323 port address, e.g., (1720).

♦ **Syntax Notes** ♦

H.323 display name string must be specified before the voice switching daughtercard is activated. A non-null string value is required.

To use this command, the **h.323 gatekeeper mode** command for gatekeeper discovery must be set to MANUAL. At least one endpoint (or gateway), and one H.323 transport address must be specified to make an over-the-network call.

Default:
None

Command Example:
voice destination to_vsd1 h.323 endpoint 225.0.1.41
voice destination to_vsd2 h.323 endpoint 225.0.1.42 1720

Remarks

This command can be used to specify the address of the gatekeeper in the currently active (H.323) zone when configured for manual mode. When the destination is a gatekeeper, port 1719 should be specified (default).

When port 1720 is used only the IP address needs to be specified.

The endpoint name string is the user's logical name for the remote calling party, and is automatically used to create the card level destinations, e.g., **tosalem**. The endpoint name string is sometimes referred to as an alias name.

voice destination local channel

Command Usage

Create a local channel destination with specified name.

Syntax Options

voice destination < *"channelDestName"* > **local channel**
< *"endpointDestName"/port/startChannel-endChannel* >

Definitions:

channelDestName	Identifies the voice call endpoint destination by name, (e.g., **tosalem**); maximum length of 40 characters. The following characters are permitted in the call destination name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & ǀ / \ < > ( ) [ ] { }
endpointDestName	Identifies the voice call endpoint destination string name unique across the H.323 network, (e.g., **to_vsd1**) any string up to 64 bytes (not including quotes). The following characters are permitted in the destination string name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & ǀ / \ < > ( ) [ ] { }
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
startChannel	The first number in the range of voice channels (e.g., **1**).
endChannel	The last number in the range of voice channels (e.g., **30**).

♦ **Syntax Note** ♦

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

Default:
None

Command Example:
voice destination tosalem local channel to_vsd1 /1/1-12
voice destination tocalab local channel to_vsd2/1/13-24

Remarks

This command is used to specify a local VSM-based gateway (VSD, VSA and VSB) destination at the channel level and add it to the list of destinations. The endpoint name string is an H.323 endpoint destination as defined in the **H.323 endpoint destination** command.

voice no destination

Command Usage

Delete an H.323 endpoint or local channel destination with specified name.

Syntax Options

voice no destination < *"DestName"* **>**

Definitions:

DestName — Identifies either the voice call H.323 endpoint destination or local channel destination by name (e.g., **to salem**), or the maximum length of 40 characters. The following characters are permitted in the call destination name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }.

Default:
None

Command Example:
voice no destination tosalem
voice no destination to_vsd1

view voice destination

Command Usage

Display an H.323 endpoint or local channel destination with specified name.

Syntax Options

view voice destination < *"DestName"* >

Definitions:

DestName	Identifies either the voice call H.323 endpoint destination or local channel destination by name (e.g., **to salem**), or the maximum length of 40 characters. The following characters are permitted in the call destination name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }.

Default:
None

Command Example:
view voice destination tosalem
view voice destination tocalab

Screen Output

To view parameters for an H.323 voice endpoint destination, type **view voice destination** and a valid destination name, e.g., **view voice destination to VSD_1**, and then press **<Enter>**.

A screen similar to the following displays.

```
******************************************
Viewing Destination
******************************************
!
voice destination VSD_1 h.323 address 195.167.10.33 1720
!
voice destination VSD_2 h.323 address 195.167.10.34 1720
!
voice destination to VSD_1 port 1 local channel VSD_1/1/1-24
!
voice destination to VSD_2 port 1 local channel VSD_1/1/1-24
!
voice phone group Ext. of PBX__1
!
voice phone group Ext. of PBX__2
!
voice phone group Ext. of PBX__1 type local extensions
!
voice phone group Ext. of PBX__2 type local extensions
!
voice phone group Ext. of PBX__1 site prefix off
!
voice phone group Ext. of PBX__2 site prefix off
!
voice phone group Ext. of PBX__1 format "xxxx"
!
voice phone group Ext of PBX_2 format "xxxx"
!
voice phone group Ext. of PBX__1 strip digit length 0
!
voice phone group Ext. of PBX__2 strip digit length 0
```

voice phone group

Command Usage

Create phone group with specified name to add to phone group list.

Syntax Options

voice phone group < *"PhoneGroupName"* >

Definitions:

PhoneGroupName — Identifies the phone group by name, (e.g., **salem_engr1**); maximum length of characters **60**. The following characters are permitted in phone group name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }.

Default:
None

Command Example:
voice phone group salem_engr1
voice phone group calab_engr2

voice no phone group

Command Usage

Delete phone group with specified name from phone group list.

Syntax Options

voice no phone group < *"PhoneGroupName"* **>**

Definitions:

PhoneGroupName — Identifies the phone group by name, (e.g., **salem_engr1**); maximum length of characters **60**. The following characters are permitted in phone group name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }.

Default:
None

Command Example:
voice no phone group salem_engr1
voice no phone group calab_engr2

view voice phone group

Command Usage

Display phone group with specified name in phone group list.

Syntax Options

view voice phone group < *"PhoneGroupName"* >

Definitions:

PhoneGroupName	Identifies the phone group by name, (e.g., **salem_engr1**); maximum length of characters **60**. The following characters are permitted in phone group name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & I / \ < > ( ) [ ] { }.

Default:
None

Command Example:
view voice phone group salem_engr1
view voice phone group calab_engr1

Screen Output

To view parameters for a voice phone group, type **view voice phone group** and a valid phone group name, e.g., **view voice phone group PBX_1**, and then press **<Enter>**.

A screen similar to the following displays.

```
*******************************************
Viewing Phone Groups
*******************************************
!
voice phone group "Ext. of PBX_1
voice phone group "Ext. of PBX_1 type local extensions
voice phone group "Ext. of PBX_1 site prefix off
voice phone group "Ext. of PBX_1 format "xxxx"
voice phone group "Ext. of PBX_1 strip digit length 0
!
```

voice phone group site prefix

Command Usage

Set unique phone group site *prefix* for routing VoIP calls (on/off).

Syntax Options

voice phone group < *“PhoneGroupName”* > site prefix {on | off}

Definitions:

PhoneGroupName	Identifies the phone group by name, (e.g., **salem_engr1**); maximum length of characters **60**. The following characters are permitted in the phone group name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
on	Turns ON site prefix for specified voice phone group.
off	Turns OFF site prefix for specified voice phone group.

Default:
The default setting is **off**.

Command Example:
voice phone group salem_engr1 site prefix off
voice phone group salem_engr2 site prefix on

Remarks

Site prefix is similar to a trunk group.

voice phone group site prefix digits

Command Usage

Specify unique phone group site *prefix digits* for routing VoIP calls.

Syntax Options

voice phone group <*"PhoneGroupName"*> site prefix digits <*"string"*>

Definitions:

PhoneGroupName — Identifies the phone group by name, (e.g., **salem_engr1**); maximum length of characters **60**. The following characters are permitted in the phone group name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

string — Identifies the phone group prefix string, (e.g., **81**); maximum length of 23 characters. The following characters are permitted in the phone group prefix string: 0-9, # and *; at least one character must be used.

♦ **Syntax Notes** ♦

If a terminating digit has been configured for the voice switching daughtercard via the **voice daughter card termination digit** command, the specified terminating digit cannot be used as a site prefix digit.

To use this command, the **voice phone group site prefix** must be turned ON.

To use this command, the associated voice numbering plan must first be activated via the **voice numbering plan activate** command.

Default:
None

Command Example:
voice phone group salem_engr1 site prefix digits ""**
voice phone group salem_engr2 site prefix digits 81

Remarks

The example in this command for the string portion of the syntax sets the site prefix digits to be used to **81**. This means that from any channel on the network, whenever the digits **81**are pressed, a certain number of digits is expected to follow. If the site prefix is set to **81**, and the format is **xxx** to get to extension **306**, for instance, the caller would dial **81306** to get to extension **306** from anywhere in the VoIP network. In this case, a two digit dialing prefix is used, so the VoIP network can support 999 sites, with **100** extensions per site.

A prefix is an indicator consisting of one or more digits allowing selection of different types of number formats (e.g., local, national or international), transit networks, and/or the service.

Prefixes are not part of the number and are not signaled over internetwork or international boundaries. When prefixes are used, the user or automatic calling equipment always enters them. Prefixes are only used on the source voice switching daughtercard, and are never sent to the remote end destination. H.323 gatekeepers do not receive prefixes as part of the alias name.

If the phone group dialing type is either NANP extensions or International extensions, then the first digit of the site prefix digit cannot be the same as the first digit of the extension, as specified via the **phone group numbers to include in range of digits** command.

voice phone group type

Command Usage

Specify phone group dialing type numbering scheme as either intra-VoIP network extensions or PSTN telephone numbers.

Syntax Options

voice phone group <*“PhoneGroupName”*> type { local extensions | nanp extensions | international extensions | nanp pstn | international pstn}

Definitions:

PhoneGroupName	Identifies the phone group by name, (e.g., **salem_engr1**; maximum length of characters **60**. The following characters are permitted in the phone group name: a-z, A-Z, 0-9, space and # * ~ ' ‘ ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
local extensions	Indicates use of intra-VoIP network (PBX) extensions between 1 and 11 digits, (e.g., **xxxx**).
nanp extensions	Indicates use of North American Numbering Plan (NANP) intra-VoIP network (PBX) extensions, (e.g., **x-xxx-xxx-xxxx**); consists of single-digit long distance designator, three-digit area code, three-digit (CO) exchange prefix, and unique four-digit code for the telephone subscriber. Any combination of three digits may now be used for the area code. NANP is the numbering scheme used to assign area codes and also to establish rules for call routing in Canada and the United States.
international extensions	Indicates use of international intra-VoIP network (PBX) extensions, (e.g., **xx-xxx-xxx-xxxx**); consists of between 10 and 26 digits (includes field separators, i.e., hyphens; without field separators maximum extension length is 15 digits); 1-3 digit country code (CC), national destination code (NDC), and subscriber number (SN). (Field separators *not available this release.*)
nanp pstn	Indicates use of North American Numbering Plan via the Public Switching Telephone Network; consists of local and long distance domestic telephone numbers, (e.g., **1-xxx-xxx-xxxx**).
international pstn	Indicates use of international long distance telephone numbers via the PSTN, (e.g., **01-xxx-xxx-xxxx**).

♦ Syntax Notes ♦

Restrictions on numbering schemes from various telephone companies are beyond the scope of this document.

More detailed, syntax-related rules specific to using local, NANP, international extensions, and NANP PSTN and International PSTN dialing types are detailed below.

Default:
The default setting is **local extensions**.

Command Example:

voice phone group salem_engr1 type local extensions 1300-1400
voice phone group salem_engr2 type nanp extensions 1-818-123-4567
voice phone group salem_engr3 type international extensions 01-234-555-6677
voice phone group calab_engr1 type nanp pstn 1-818-123-4567
voice phone group caleb_engr2 type international pstn 01-234-555-6677

Syntax Notes

Local Extensions (intended for daughtercard to daughtercard calls)

- If local extensions used, then voice switching daughtercard recognizes *fixed* number of dialed extension digits, i.e., **0, 1, 2, 3, 4, 5, 6, 7, 8, 9, 10** or **11**. *No site prefix required.*
- If H.323 gatekeeper control is set to ON via the **h.323 gatekeeper control** command, then the telephone extensions are set via the phone group **numbers to include in range of digits** command, and automatically registered with the gatekeeper as an alias.
- Format string must have between one and seven numbering scheme place-holders or designators (**xxxxxxx**); **0** through **9** are allowed in any combination as long as at least one "**x**" is used, e.g., **1x, 2xx, 1xxxxxx, 2xxxxxx,** etc. A dialed digit cannot follow an "**x**". Neither the # and * symbols nor letters A through E are allowed. Format string can only contain dialed digits (no field separators).
- Local extensions cannot be used in conjunction with any other dialing types.

NANP Extensions (intended for daughtercard to daughtercard calls)

- If NANP extensions used digital voice switching daughtercard recognizes *variable* number of dialed digits, i.e., between seven and 11 digits; 11 digits when long distance designator included.
- If H.323 gatekeeper control is set to ON via the **h.323 gatekeeper control** command, then the telephone extensions are set via the phone group **numbers to include in range of digits** command, and automatically registered with the gatekeeper as an alias.
- If long distance designator used it can only contain one digit which must be the number **1** followed by field separator. (Field separators *not available this release.*)
- Three-digit central office (CO) code, field separator and unique four-digit subscriber number are required.
- Format string must contain dialed digits and limited use of field separators. Field separators (hyphens) in sequence indicate empty field; two field separators are not allowed in sequence as part of a valid format string. A dialed digit cannot follow an "**x**". At least one "**x**" is required.

  If no format is specified, then the phone group **numbers to include in range of digits** command is not allowed. Command can use only **1, 2, 3, 4,** or **7** numbering scheme "**x**" designators. Designators are not allowed in area code or prefixes unless long distance designator is used, in which case three designators must be used in the prefix (but never the area code), e.g., **818-123-000x, 1-818-xxx-xxxx.** Allowed digits include **000** through **999.**

International Extensions (intended for daughtercard to daughtercard calls)

- If NANP extensions used then voice switching daughtercard recognizes *variable* number of dialed digits, i.e., between 10 and 26 digits; 26 digits when long distance designator(s) and site prefix included. (See international PSTN extensions for more information on site prefixes.)
- If H.323 gatekeeper control is set to ON via the **h.323 gatekeeper control** command, then the telephone extensions are set via the phone group **numbers to include in range of digits** command, and automatically registered with the gatekeeper as an alias.
- Format string must contain dialed digits and limited use of field separators. Field separators (hyphens) in sequence indicate empty field; two hyphens are not allowed in sequence as part of a valid format string. A dialed digit cannot follow an "x". At least one "x" is required.
- Format is optional; if not specified, the phone group **numbers to include in range of digits** command is not allowed.

NANP PSTN (intended for daughtercard to PSTN calls)

- If NANP PSTN telephone numbers used then voice switching daughtercard recognizes *variable* number of dialed digits, i.e., between seven and 11 digits; *11 digits when long distance designator included.*
- If H.323 gatekeeper control is set to ON via the **h.323 gatekeeper control** command, then the telephone numbers are *not* registered with the gatekeeper as an alias.
- Format string must contain dialed digits and limited use of field separators. Field separators (two hyphens) in sequence indicate empty field; two field separators are not allowed in sequence as part of a valid format string. A dialed digit cannot follow an "x". At least one "x" is required.
- Format is optional; if not specified, the phone group **numbers to include in range of digits** command is not allowed.

PSTN International (intended for daughtercard to PSTN calls)

- If PSTN international telephone numbers used then voice switching daughtercard recognizes *variable* number of dialed digits, i.e., between 10 and 26 digits; *26 digits when long distance designator(s) and required site prefix included.*

  Length of site prefix digits, which can be any combination of numbers from 0 to 9, cannot exceed 26 digits total when added to the format string. Site prefix digits are set via the **voice phone group site prefix digits** and **voice phone group site prefix (on/off)** commands.
- One- to three-digit country code (CC), three-digit central office (CO) code, field separator and unique four-digit subscriber number (SN) are required; cannot exceed 15 digits when field separators not used.
- Format string must contain dialed digits and limited use of field separators. Field separators (hyphens) in sequence indicate empty field; two field separators are not allowed in sequence as part of a valid format string. A dialed digit cannot follow an "x". At least one "x" is required.

  Format must be compatible with ITU E.164.1 specification.
- Format is optional; if not specified, the phone group **numbers to include in range of digits** command is not allowed.

voice phone group format

Command Usage

Specify phone group format of telephone number and number of outbound digits to dial.

Syntax Options

voice phone group < *"PhoneGroupName"* > **format** < *"formatString"* >

Definitions:

PhoneGroupName — Identifies the phone group by name, (e.g., **salem_engr1**); maximum length of characters **60**. The following characters are permitted in the phone group name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

formatString — Identifies the phone group format string, (e.g., **xxx**); maximum length of characters **24**. The only character permitted in the phone group format string: x and X.

♦ **Syntax Notes** ♦

Valid field separators include ( ) , . = _ + [ ] { } \ / : ; < or space (no hyphens); field separators are not allowed for local extensions.

For this command only, a dialed digit can be **0, 1, 2, 3, 4, 5, 6, 7, 8, 9** and **x**.

For the format string, a dialed digit cannot follow an "**x**" designator there must be at least one "**x**" (up to seven placeholder or designators allowed).

Format is optional; if format specified, the phone group **numbers to include in range of digits** command is not allowed. Also, if format specified, then site prefix digits must be set and turned ON via the **voice phone group site prefix digits** and **voice phone group site prefix (on/off)** commands.

If a termination digit has been set via the **VSD termination digit** command, then that digit cannot be used in this command.

The digits specified in the site prefix digits and the number of digits implied by the format must be unique on the entire network numbering plan for different types.

Default:
None

Command Example:
voice phone group salem_engr1 format 1xxx
voice phone group salem_engr2 format 2xxx
voice phone group salem_engr3 format 81x xxxx
voice phone group calab_engr1 format 818 xxxx
voice phone group calab_engr2 format 31xx
voice phone group calab_engr3 format 41xx

Remarks

This command calculates the number of digits allowed and describes the flexible portion of a dialing number to collect.

voice phone group strip digit length

Command Usage

Specify number of outbound (collected) digits to strip in phone group before forwarding call (optional).

Syntax Options

voice phone group < *"PhoneGroupName"* **> strip digit length <***num* **>**

Definitions:

PhoneGroupName	Identifies the phone group by name, (e.g., **salem_engr1**); maximum length of 40 characters. The following characters are permitted in the phone group name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
num	Identifies the phone group strip digit length string, (e.g., **2**); maximum string length is **24**; minimum is 0.

Default:
The default *num* value is **0**.

Command Example:
voice phone group salem_engr1 strip digit 0
voice phone group salem_engr2 strip digit 1
voice phone group salem_engr3 strip digit 2

voice phone group forwarding prefix

Command Usage

Specify string of digits to prefix before fowarding call to endpoint destination.

Syntax Options

voice phone group <*“PhoneGroupName”*> forwarding prefix {on | off}

Definitions:

PhoneGroupName	Identifies the phone group by name, (e.g., **salem_engr1**); maximum length of 40 characters. The following characters are permitted in the phone group name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
on	Turns ON forwarding of voice phone group prefix.
off	Turns OFF forwarding of voice phone group prefix.

Default:
The default setting is **off.**

Command Example:
voice phone group salem_engr1 forwarding prefix off
voice phone group salem_engr2 forwarding prefix on

voice phone group forwarding prefix digits

Command Usage

Specify voice phone group digits to prefix before forwarding call (optional).

Syntax Options

voice phone group < *"PhoneGroupName"* > **forwarding prefix digits** < *"PrefixNum"* >

Definitions:

PhoneGroupName — Identifies the phone group by name, (e.g., **salem_engr1**); maximum length of 40 characters. The following characters are permitted in the phone group name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

PrefixNum — Identifies the forwarding prefix digits in the specified phone group, (e.g., **9**); maximum string length is **14** (MAX_DIAL_DIGITS-1); each digit can be either 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, or #.

♦ **Syntax Notes** ♦

To use this command, the **phone group forwarding prefix** command must first be enabled.

Default:

The default *PrefixNum* value is **0**.

Command Example:

voice phone group salem_engr1 strip digit 0
voice phone group salem_engr2 strip digit 1
voice phone group salem_engr3 strip digit 2

voice phone group

Command Usage

Set type of access calls (voice, fax, modem, data) allowed in phone group (on/off). (*Not available this release.*)

Syntax Options

voice phone group <*"PhoneGroupName"*> **usage {voice | usage fax | usage modem | usage data} {on | off}**

Definitions:

PhoneGroupName	Identifies the phone group by name, (e.g., **salem_engr1**); maximum length of characters **60**. The following characters are permitted in the phone group name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }
on	Turns ON voice, fax, modem or data for specified voice phone group.
off	Turns OFF voice, fax, modem or data for specified voice phone group.

Default:
The default setting is **off**.

Command Example:
voice phone group salem_engr1 usage voice off
voice phone group salem_engr2 usage fax on
voice phone group calab_engr1 usage modem off
voice phone group calab_engr2 usage data on

voice phone group add numbers

Command Usage

Specify phone group *numbers to include* in range of digits for phone format string.

Syntax Options

voice phone group < *"PhoneGroupName"* > **add numbers** < *"StartRange"* > **[[thru]** *"EndRange"*]

Definitions:

PhoneGroupNam	Identifies the phone group by name, (e.g., **salem_engr1**); maximum length of 40 characters. The following characters are permitted in the phone group name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }.
StartRange	Specifies starting digits to add to designated phone group as per format specified in **voice phone group format string** command, (e.g., extension **4600**).
thru	Optional command syntax.
EndRange	Specifies ending digits to add to designated phone group as per format specified in **voice phone group format string** command, (e.g., extension **4800**).

♦ **Syntax Notes** ♦

Ranges must be unique across the entire network numbering scheme, and digits specified must match the allowed number of digits as specified in the format string via the **voice phone group format string** command.

Termination digits as specified via the **VSD termination digit** command cannot be specified in either range.

Multiples of this command can be issued to have a cumulative effect.

If no number is entered for the end range, then the start range value is used.

If no number is entered for the starting range, all zeroes will be used based upon the specified format string.

Default:
None

Command Example:

voice phone group salem_engr1 add numbers 4600 thru 4800
voice phone group salem_engr1 add numbers 4600 4800
voice phone group calab_engr1 add numbers 2500 thru 2750
voice phone group calab_engr1 add numbers 2500 2750

Remarks

If the phone group dialing type is NANP extensions or International extensions, then the first digit of the site prefix digits cannot be the same as the first digits as specified via the **phone group numbers to include in range of digits** command.

voice phone group delete numbers

Command Usage

Specify phone group *numbers to remove* from range of digits for phone format string. (*Not available this release.*)

Syntax Options

voice phone group < *"PhoneGroupName"* > **delete numbers** < *"StartRange"* > **[[thru]** *"EndRange"* **]**

Definitions:

PhoneGroupName	Identifies the phone group by name, (e.g., **salem_engr1**); maximum length of 40 characters. The following characters are permitted in the phone group name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }.
StartRange	Specifies starting digits to delete from designated phone group as per format specified in **voice phone group format string** command, (e.g., four-digit local extension **4600**).
thru	Optional command syntax.
EndRange	Specifies ending digits to delete from designated phone group as per format specified in **voice phone group format string** command, (e.g., four-digit local extension **4800**).

Default:
None

Command Example:
voice phone group salem_engr1 delete numbers 4600 thru 4800
voice phone group salem_engr1 delete numbers 4600 4800
voice phone group calab_engr1 delete numbers 2500 thru 2750
voice phone group calab_engr1 delete numbers 2500 2750

voice numbering plan

Command Usage

Create numbering plan with specified name.

Syntax Options

voice numbering plan < *"NumberingPlanName"* **>**

Definitions:

NumberingPlanName Identifies the numbering plan by name, (e.g., **salem#plan1**); maximum length of 40 characters. The following characters are permitted in the numbering plan name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }.

♦ Syntax Notes ♦

If the H.323 **gatekeeper control** command is ON, then the voice switching daughtercard automatically generates an additional alias name when the **activate voice numbering plan** command is issued.

Numbering plan names cannot be named "all" because "all" is a reserved numbering plan name.

Numbering plans do not take effect unless they are activated via the **activate numbering plan** command.

Default:
None

Command Example:
voice numbering plan salem#plan1
voice numbering plan calab#plan2

Remarks

Numbering plans associate one or more groups to one or more destinations (hunting targets) to be called.

voice no numbering plan

Command Usage

Delete numbering plan with specified name.

Syntax Options

voice no numbering plan < *"NumberingPlanName"* **>**
<u>Definitions</u>: *NumberingPlanName* Identifies the numbering plan by name, (e.g., **salem#plan1**); maximum length of 40 characters. The following characters are permitted in the numbering plan name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }. <u>Default</u>. None <u>Command Example</u>: **voice no numbering plan salem#plan1** **voice no numbering plan calab#plan2**

view voice numbering plan

Command Usage

Delete numbering plan with specified name.

Syntax Options

view voice numbering plan <*"NumberingPlanName"*>

Definitions:

NumberingPlanName Identifies the numbering plan by name, (e.g., **salem#plan1**); maximum length of 40 characters. The following characters are permitted in the numbering plan name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }.

Default:
None

Command Example:
view voice numbering plan salem#plan1
view voice numbering plan calab#plan2

Screen Output

To view a voice numbering plan, type **view voice numbering plan** followed by a valid numbering plan name, e.g., **view voice numbering plan PBX_1**, and then press **<Enter>**.

A screen similar to the following displays.

```
******************************************
Viewing Numbering Plan
******************************************
!
voice numbering plan to PBX_1
!
voice numbering plan to PBX_2
!
voice numbering plan to PBX_1 hunt method round robin
!
voice numbering plan to PBX_1 hunt method round robin
!
voice numbering plan to PBX_1 associate destination member to VSD_1
!
voice numbering plan to PBX_2 associate destination member to VSD_2
!
voice numbering plan to PBX_1 associate phone group member Ext. of PBX_1
!
voice numbering plan to PBX_2 associate phone group member Ext. of PBX_2
!
voice numbering plan to PBX_1 description trunk to route calls from VSD1 to PBX1
!
voice numbering plan to PBX_2 description trunk to route calls from VSD2 to PBX2
!
voice numbering plan to PBX_1
!
```

voice numbering plan activate

Command Usage

Activate voice numbering plan with specified name. (*Not available this release.*)

Syntax Options

voice numbering plan {all | activate *"NumberingPlanName"* **}**

Definitions:

all — Activates all numbering plans at once.

activate — Activates only specified numbering plan.

NumberingPlanName — Identifies the numbering plan by name, (e.g., **salem#plan1**); maximum length of 40 characters. The following characters are permitted in the numbering plan name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }.

♦ Syntax Notes ♦

If the H.323 **gatekeeper control** command is ON, when the **activate voice numbering plan** command is issued, the voice switching daughtercard automatically generates an additional alias name.

Default:
None

Command Example:
voice numbering plan all
voice numbering plan activate salem#plan1
voice numbering plan activate calab#plan2

Remarks

Numbering plans do not take effect until this command is issued.

Once a numbering plan is activated, all new connections are temporarily halted until this command is completed. It can take up to approximately 10 seconds for a numbering plan to be activated. As a result, it is recommended that this command be issued only when it will have minimum impact on callers. It is also more efficient to activate all the numbering plans at once, rather than individually.

voice numbering plan hunt method

Command Usage

Specify numbering plan method of outgoing hunting (destination group).

Syntax Options

voice numbering plan < *"NumberingPlanName"* > hunt method {round robin | top down}

Definitions:

NumberingPlanName	Identifies the numbering plan by name, (e.g., **salem#plan1**); maximum length of 40 characters. The following characters are permitted in the numbering plan name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }.
round robin	Indicates numbering plan uses round robin hunting method to find open lines for incoming calls.
top down	Indicates numbering plan uses top down hunting method to find open lines for incoming calls.

Default:
The default setting is **round robin**.

Command Example:
voice numbering plan salem#plan1 hunt method round robin
voice numbering plan calab#plan2 hunt method top down

Remarks

This command also groups related destinations together.

The round robin hunting method starts from the destination member just after the last used destination member each time a hunt request is received. The last used destination member is "remembered" across sessions.

The top down hunting method starts from the first destination member in the hunt group each time a new session is started.

Hunt groups allow telephone lines to be organized so that when the first line tried is unavailable for an incoming call, the next available line (using either the round robin or top down method) is hunted until an open line is located.

voice numbering plan description

Command Usage

Define numbering plan of specified phone group (optional).

Syntax Options

voice numbering plan < *"NumberingPlanName"* > **description** < *"string"* >

Definitions:

NumberingPlanName	Identifies the numbering plan by name, (e.g., **salem#plan1**); maximum length of 40 characters. The following characters are permitted in the numbering plan name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }.
string	Text string that describes the specified voice numbering plan, (e.g., **phone group a_eng**); maximum length of characters **40**. The following characters are permitted in the numbering plan description string: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }.

Default:
None

Command Example:
voice numbering plan salem#plan1 description phone group a_eng
voice numbering plan salem#plan2 description phone group b_eng
voice numbering plan calab#plan1 description phone group a_eng
voice numbering plan calab#plan2 description phone group b_eng

Remarks

This command is used to store a description of this phone group for convenience. Because the command is optional it has no effect on the switch. It can also be used to hold the circuit identifier (see the **voice port circuit identifier** command).

voice numbering plan destination member

Command Usage

Associate (or disassociate) destinations, or hunting targets, with numbering plans by name.

Syntax Options

voice numbering plan < *"NumberingPlanName"* > **{associate | disassociate} destination member** < *"DestName"* >

Definitions

NumberingPlanName	Identifies the numbering plan by name, (e.g., **salem#plan1**); maximum length of 40 characters. The following characters are permitted in the numbering plan name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }.
associate	Indicates specified numbering plan associated with destination member.
disassociate	Indicates specified numbering plan disassociated with destination member
DestName	Identifies either the voice call H.323 endpoint destination or local channel destination by name, (e.g., **to salem**) or the c; maximum length of 40 characters. The following characters are permitted in the call destination name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }.

♦ **Syntax Notes** ♦

To use this command, the hunt method must first be specified via the **numbering plan outgoing hunt method** command.

It is recommended that a local channel destination be used for the endpoint name string.

Default:
The default setting is **associate.**

Command Example:
voice numbering plan salem#plan1 associate destination member tocalab
voice numbering plan salem#plan2 disassociate destination member to_vsd1
voice numbering plan calab#plan1 associate destination member tosalem
voice numbering plan calab#plan2 disassociate destination member to_vsd2

Remarks

This command is used to append a destination (hunting target) to the associated numbering plan for telephone number hunting. The numbering plan destination list is used to hunt for destinations (targets). Each numbering plan destination list requires at least one member.

voice numbering plan phone group member

Command Usage

Associate (or disassociate) phone groups with numbering plans by name.

Syntax Options

voice numbering plan < *"NumberingPlanName"* > **{associate | disassociate} phone group member** < *"PhoneGroupName"* >

Definitions:

NumberingPlanName	Identifies the numbering plan by name, (e.g., **salem#plan1**); maximum length of 40 characters. The following characters are permitted in the numbering plan name: a-z, A-Z, 0-9, space and # * ~ ' ` ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }.
associate	Indicates specified phone group associated with specified numbering plan.
disassociate	Indicates specified phone group disassociated with specified numbering plan
PhoneGroupName	Identifies the phone group by name, (e.g., **calab_engr1**); maximum length of characters **60**. The following characters are permitted in the phone group name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , .@ $ % ^ _ & | / \ < > ( ) [ ] { }

♦ **Syntax Notes** ♦

To use this command, the destination member must first be associated via the **associate destination member** command.

At least one phone group must be associated before activating the associated numbering plan.

Default:
None

Command Example:
voice numbering plan salem#plan1 associate phone group member calab_engr1
voice numbering plan salem#plan2 disassociate phone group member calab_engr2
voice numbering plan calab#plan1 associate phone group member salem_engr1
voice numbering plan calab#plan2 disassociate phone group member salem_engr2

System-Wide VoIP Commands

The commands listed and described below are used to display system-wide VoIP command settings and statistics as follows: various VSD level parameters and configured voice items, including telephony, telephony channel, voice play out, dsp (receive and transmit), errors, modem, fax and ISDN statistics.

View Voice Switching Daughtercard Parameters

View Voice Switching Daughtercard Port Parameters

View Voice Switching Daughtercard Channel Parameters

View Voice Switching Daughtercard Network Parameters

View Statistics

telephony statistics
channel statistics
voice play out statistics
dsp receive and transmit statistics
error statistics
modem statistics
fax statistics
ISDN level 2 statistics

view voice daughter card

Command Usage

Display voice switching daughtercard parameters at the daughtercard level.

Syntax Options

view voice daughter card <*slot/card_number*>

Definitions:

slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
card_number	Specifies physical port number on voice daughtercard, (e.g., **1**).

Default:
None

Command Example:
view voice daughter card 2/1
view voice daughter card 2/2
view voice daughter card 3/1
view voice daughter card 3/2

Screen Output

To view voice daughtercard parameters, type **view voice daughter card** followed by valid slot and daughtercard port numbers, e.g., **view voice daughtercard 4/1**, and then press **<Enter>**.

A screen similar to the following displays.

```
********************************************
Viewing Daughter Card
********************************************
!
voice daughtercard 4/1 ip address 127.0.0.0
!
voice daughtercard 4/1 ip mask 255.255.255.0
!
voice daughtercard 4/1 activate
```

view voice port

Command Usage

Display voice switching daughtercard parameters at the port level.

Syntax Options

view voice port <*slot/port*>

Definitions:

slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).

Default:
None

Command Example:
view voice port 2/1
view voice port 2/2
view voice port 2/3
view voice port 2/4

Screen Output

To view voice daughtercard port parameters, type **view voice port** followed by valid slot and daughtercard port numbers, e.g., **view voice port 4/1**, and then press **<Enter>**.

A screen similar to the following displays.

```
*******************************************
Viewing Port
*******************************************
!
voice port 4/1 interface type T1
!
```

view voice channel

Command Usage

Display voice switching daughtercard parameters at the channel level.

Syntax Options

view voice channel <*slot/port/channel*>

Definitions.

slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
channel	Specifies port channel number in which to view voice channel parameters, (e.g., **25**).

Default:
None

Command Examples:
view voice channel 2/1/1
view voice channel 2/1/15
view voice channel 2/1/17
view voice channel 2/1/30
view voice channel 2/1/60
view voice channel 2/2/1
view voice channel 2/2/15
view voice channel 2/2/17
view voice channel 2/2/30

Screen Output

To view voice daughtercard parameters, type **view voice channel** followed by valid slot, daughtercard port and channel number(s), e.g., **view voice channel 4/1/1-12**, and then press **<Enter>**.

A screen similar to the following displays.

```
*******************************************
Viewing Channel
*******************************************
!
voice channel 4/1/1 mode telephony
voice channel 4/1/2 mode telephony
voice channel 4/1/3 mode telephony
voice channel 4/1/4 mode telephony
voice channel 4/1/5 mode telephony
voice channel 4/1/6 mode telephony
voice channel 4/1/7 mode telephony
voice channel 4/1/8 mode telephony
voice channel 4/1/9 mode telephony
voice channel 4/1/10 mode telephony
voice channel 4/1/11 mode telephony
voice channel 4/1/12 mode telephony
```

!

view voice network card

Command Usage

Display voice switching daughtercard parameters at the network level.

Syntax Options

view voice network card <*slot/card_number*>

Definitions:

slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
card_number	Specifies physical port number on voice switching daughtercard, (e.g., **1**).

Default:
None

Command Example:
view voice network card 2/1
view voice network card 2/2
view voice network card 3/1
view voice network card 3/2

Screen Output

No screen output available at this time.

view voice channel telephony level stats

Command Usage

Display telephony statistics.

Syntax Options

view voice channel <*slot/port/channel*> **telephony level stat[istic]s**

Definitions:

slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
channel	Specifies port channel number in which to view telephony level statistics, (e.g., **25**).
istic	Optional command syntax. You can type either **stats** or **statistics** in the command line.

Default:
None

Command Example:
view voice channel 2/1 25 telephony level statistics
view voice channel 2/1 25 telephony level stats
view voice channel 2/2 30 telephony level statistics

Remarks

Displays the current receive levels, mean receive levels, transmit levels and the mean transmit levels at the PCM interface of the DSP for the specified channel (received from or transmitted to the telephony interface). Current levels are given in 0.1 dBm0 units. Mean values are in 0.1 linear PCM units.

view voice channel telephony channel stats

Command Usage

Display channel statistics.

Syntax Options

view voice channel <*slot/port/channel*> **telephony channel stat[istic]s**

Definitions:

slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
channel	Specifies port channel number in which to view cumulative telephony channel statistics, (e.g., **25**); cumulative since channel was placed in service.
istic	Optional command syntax. You can type either **stats** or **statistics** in the command line.

Default:
None

Command Example:
view voice channel 2/1 25 telephony level statistics
view voice channel 2/1 25 telephony level stats
view voice channel 2/2 30 telephony level statistics
view voice channel 2/2 30 telephony level stats

view voice channel voice playout stats

Command Usage

Display DSP voice play out statistics.

Syntax Options

view voice channel <*slot/port/channel*> **voice playout stat[istic]s**

Definitions:

slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
channel	Specifies port channel number in which to view DSP voice play out statistics, (e.g., **25**); cumulative for the current call on specified channel.
istic	Optional command syntax. You can type either **stats** or **statistics** in the command line.

Default:
None

Command Example:
view voice channel 2/1 25 voice playout statistics
view voice channel 2/1 25 voice playout stats
view voice channel 2/2 30 voice playout statistics
view voice channel 2/2 30 voice playout stats

view voice channel dsp stats

Command Usage

Display DSP voice play out statistics.

Syntax Options

view voice channel <*slot/port/channel*> **dsp stat[istic]s**

Definitions:

slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
channel	Specifies port channel number in which to view DSP receive and transmit statistics, (e.g., **25**); cumulative for the current call on specified channel.
istic	Optional command syntax. You can type either **stats** or **statistics** in the command line.

Default:
None

Command Example:
view voice channel 2/1 25 dsp statistics
view voice channel 2/1 25 dsp stats
view voice channel 2/2 30 dsp statistics
view voice channel 2/2 30 dsp stats

view voice channel error stats

Command Usage

Display error statistics.

Syntax Options

view voice channel <*slot/port/channel*> **error stat[istic]s**

Definitions:

slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
channel	Specifies port channel number in which to view error statistics, (e.g., **25**); cumulative for the current call on specified channel.
istic	Optional command syntax. You can type either **stats** or **statistics** in the command line.

Default:
None

Command Example:
view voice channel 2/1 25 error statistics
view voice channel 2/1 25 error playout stats
view voice channel 2/2 30 error playout statistics
view voice channel 2/2 30 error playout stats

view voice channel modem stats

Command Usage

Display modem statistics.

Syntax Options

view voice channel <*slot/port/channel*> modem stat[istic]s

Definitions:

slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
channel	Specifies port channel number in which to view modem statistics, (e.g., **25**); cumulative for the current call on specified channel.
istic	Optional command syntax. You can type either **stats** or **statistics** in the command line.

Default:
None

Command Example:
view voice channel 2/1 25 modem statistics
view voice channel 2/1 25 modem stats
view voice channel 2/2 30 modem statistics
view voice channel 2/2 30 modem stats

view voice channel fax stats

Command Usage

Display facsimile statistics.

Syntax Options

view voice channel <*slot/port/channel*> **fax stat[istic]s**

Definitions:

slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
channel	Specifies port channel number in which to view facsimile statistics, (e.g., **25**); cumulative for the current call on specified channel.
istic	Optional command syntax. You can type either **stats** or **statistics** in the command line.

Default:
None

Command Example:
view voice channel 2/1 25 fax statistics
view voice channel 2/1 25 fax stats
view voice channel 2/2 30 fax statistics
view voice channel 2/2 30 fax stats

view voice channel isdn level 2 stats

Command Usage

Display ISDN level 2 statistics.

Syntax Options

view voice channel <*slot/port/channel*> isdn level 2 stat[istic]s

Definitions:

slot	Specifies chassis slot number where VSM is installed, (e.g., **2**).
port	Specifies physical port number on voice daughtercard, (e.g., **1**).
channel	Specifies port channel number in which to view ISDN level 2 statistics, (e.g., **25**).
istic	Optional command syntax. You can type either **stats** or **statistics** in the command line.

Default:
None

Command Example:
view voice channel 2/1 25 isdn level 2 statistics
view voice channel 2/1 25 isdn level 2 stats
view voice channel 2/2 30 isdn level 2 statistics
view voice channel 2/2 30 isdn level 2 stats